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NASA-LEWIS Research Center

Cleveland, Ohio

November 1, 1980 - October 31, 1981

NASA Grant: NAG 3-119

Donald L. Schilling

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Principal Investigator

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DEPARTMENT OF ELECTRICAL ENGINEERING

THE CITY COLLEGE OF
THE CITY UNIVERSITY of NEW YORK

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1. Introduction

1.1 The Mobile Telephone Service Market

The land mobile radio market which includes police radio, taxicab dispatch, emergency medical, and public mobile telephone is expected to be growing at the rate of about 15 percent per year. The mobile radio telephone system which is part of this market is reletively small, it is expected to grow rapidly within the next decade.

The current market is serviced by both the telephone companies and independent radio common carriers. Only the telephone companiers offer the improved mobile telephone service system in most major metropolitan areas while both the telephone companies and independent radio common carries offer manual dispatch systems in smaller cities and towns.

The cellular concept (e.g. Bell System's AMPS) has great promise for improving the mobile telephone service in the urban area. The coverage area of a cellular system is divided into small geographic cells, each of which will have its own antenna and low power frequency modulation transceiver. Calls are first

routed to a switching office and then by phone line to the appropriate cell transceiver for radio transmission to the mobile. Thus the cellular system is a terrestrial system and is economically sound only in centralized population districts. Two cellular systems are undergoing tests in Chicago and in the Baltimore-Washington area. It is expected that such systems will replace the conventional mobile telephone service systems once it is proven to be profitable and provides services of better quality.

It is currently estimated that cellular systems will cover approximately one third of the standard metropolitan statistical areas of the U.S. by the year of 1990. However, the total of metropolitan statistical areas is only about 9 percent of the geographical area of the country. This will leave a vast geographical area and about 100 million people (1970 census) without cellular service.

In these nonmetropolitan areas which are not covered by the cellular system, it may prove cost-effective to use a Land Mobile Satellite System (LMSS).

1.2 The Project

This project is the result of a joint effort between the Department of Electrical Engineering at the City College, CUNY; and the Department of Computer & Information Science at the Brooklyn College, CUNY.

The goal of this project is to develop a software simulator to help NASA in the design of the LMSS. The simulator will be used to study the characteristics and implementation requirements of the LMSS's configuration with specifications as outlined by NASA.

2. The Land Mobile Satellite System (LMSS)

2.1 The Configuration

As discussed in section 1.1, it is envisioned that the LMSS will service all mobile communication in the rural area where the population density is thinly distributed, and that terrestrial cellular systems will service all mobile communication in the urban area. The interface between the satellite system and the terrestrial system will be done through the regular wire-line telephone network. The 'total' telephone system is depicted in figure 2.1.1. The component shown are the regular wire-line telephone network, the satellite system (LMSS), and the cellular system (AMPS).

The basic LMSS to be designed by NASA is illustrated in figures 2.1.2 and 2.1.3 which were supplied by NASA. As shown in figure 2.1.2, the area of coverage will be divided into eighteen regions, each serviced by its own S-beam. In turn, the area covered by each S-beam will be serviced by a few UHF-beams.

As shown in figure 2.1.3, the LMSS will consist of the following devices: satellite, master control station, gateways,

mobile units, portable units, and private base stations.

Within each S-beam, there will be serveral service areas, which correspond to the UHF-beams. There will be a gateway in each service area to monitor the communications within the area, and one of the gateways will be chosen to be the master control station. The gateway functions include the following: directing calls, connection to the telephone network, connection to private base stations, and forwarding calls to the master control station.

The master control station will serve as the clearing house between all gateways. If a gateway has to forward a call to another gateway, the call will first be forwarded to the master control station, then the master control station will direct the call to the destination gateway.

We analyzed the LMSS and derived the system block diagram as shown in figure 2.1.4. The master control station and the gateways are connected to the telephone network which will forward calls to/accept calls from a fixed phone or a mobile serviced by the cellular systems.

Terrestrial communication lines will exist among the master control station, gateways, and private base stations. These communication lines will be used for the purpose of linking their

facilities and to provide the neccessary services, such as, forwarding calls.

The master control station, gateways, private base stations, mobile units, and portable units will all communicate with the satellite through the satellite channel which includes both the S-beams and UHF-beams. Interference sources also enter into the satellite channel and must be considered.

2.2 Types of Call

An analysis was done by NASA of the types of calls that may occur within the LMSS. The following is the resultant list:

1. a mobile wants to communicate with another mobile in the same S-beam and same UHF-beam;
2. a mobile wants to communicate with another mobile in the same S-beam but in a different UHF-beam;
3. a mobile wants to communicate with another mobile in a different S-beam;
4. a mobile in the LMSS wants to communicate to a fixed phone or another mobile serviced by a cellular system in the same S-beam;
5. a mobile in the LMSS wants to communicate to a fixed phone or another mobile serviced by a cellular system but in a different S-beam;
6. a fixed phone or a mobile serviced by a cellular system wants to communicate with a mobile in the LMSS in the same S-beam;
7. a fixed phone or a mobile serviced by a cellular system wants to communicate with a mobile in the LMSS but in a different S-beam;

They are summerized in table 2.2.1 and llustrated in figure 2.2.2.

Table 2.2.1: Types of Call

Definitions: M - rural based mobile serviced by satellite system

C - cellular based mobile serviced by cellular system

F - any phone on the fixed wire line network

Type	Call	Decription
1	M1 to M2	Rural mobile to rural mobile in same UHF-beam
2	M1 to M3	Rural mobile to rural mobile in different UHF-beam, same S-beam
3	M1 to M4	Rural mobile to rural mobile in different UHF-beam, different S-beam
4	M1 to F/C1	Rural mobile to fixed/cellular mobile in same S-beam
5	M1 to F/C2	Rural mobile to fixed/cellular mobile in different S-beam
6	F/C1 to M1	Fixed/cellular mobile to rural mobile in same S-beam
7	F/C2 to M1	Fixed/cellular mobile to rural mobile in different S-beam

2.3 Link options

NASA further studied the type of calls and determined the possible link options within each type of calls. They are summerized in table 2.3.1.

Table 2.3.1: Link Options

Class	Description	Link	Option
1	M1 to M2	1	M1->M2, hard wired transponder
		2	M1->M2, direct switched transponder
		3	M1->M2, indirect switched transponder
		4	M1->G1->M2, double hop system
2	M1 to M3	1	M1->M3, hard wired transponder
		2	M1->M3, direct switched transponder
		3	M1->M3, indirect switched transponder
		4	M1->G1->M3, double hop system
3	M1 to M4	1	M1->M4, hard wired transponder
		2	M1->M4, direct switched transponder
		3	M1->M4, indirect switched transponder
		4	M1->G1->M4, double hop, single gateway system

		5	M1->G2->M5, double hop, single gateway system
		6	M1->G1->G2->M4, double hop, double gateway system
4	M1 to F/C1	1	M1->G1->F/C1, single hop system
5	M1 to F/C2	1	M1->G1->F/C2, single hop system
		2	M1->G2->F/C2, hard wired transponder
		3	M1->G2->F/C2, switched transponder
6	F/C1 to M1	1	F/C1->G1->M1, single hop system
7	F/C2 to M1	1	F/C2->G1->M1, single hop system
		2	F/C1->G2->M1, hard wired transponder
		3	F/C1->G2->M1, switched transponder

3. The Design of the Simulator

3.1 The Overall Design of the Simulator

FORTRAN IV is the language that we chose to implement the simulator because of several reasons:

a) The most important factor is that FORTRAN IV is one the most, if not the most, portable programming language which is standardized. Almost all computer installations are capable of running a FORTRAN IV program.

b) For the simulator, many scientific calculation are performed and FORTRAN IV is one of the few languages which is scientificcally oriented.

c) Because of the number of iterations of each simulation run and the amount of work required to process a sample through the system, speed is a very important factor for designing the system. FORTRAN IV is known to be a fast compiler, and also the coding generated by the FORTRAN IV is also known to be very efficient.

The concepts of structured and modular programming were employed into the design of the system. Moreover, a 'top-down' design was used in the design of the simulation.

Top down design means the modules were designed, coded and debugged by starting from the main program of the overall system and working all the way down to the basic modules at the bottom. Figure 3.1.1 represents the tree structure of the system. The advantages of top down design are: program structure will be the same as the data flow structure, modular programming will be facilitated, and loose coupling between modules will be ensured.

Structured programming means that for any subprocedure, there can be only one way to 'ENTER' and one way to 'EXIT'. The type of statements that can be contained in the subprocedure are:

a) a block of consecutive statements which can be executed in the physical sequence as they appear;

b) conditional statement which allow the choice of executing one out of several blocks of statements, e.g. IF statement, CASE statement;

c) a repeated loop with only one mature condition to branch out of the loop which locates either at the front or at the end of

the loop;

The execution flow diagrams of these types of statements are shown in figure 3.1.2, in which all types have only one 'ENTER' and one

'EXIT', thus structured programming can be ensured. Structured programming will improve the readibility of the program and flow of execution will be on a top down fashion.

Although FORTRAN IV is not a structured language like ALGOL or PL/I, we designed the flow of execution by using structured programming techniques. This was achieved by first developing the algorithm in a highly structure manner by using an ALGOL-like pseudo language. Thus, the algorithm we use is itself language independent. After this is done, it is a simple matter to encode our algorithm intc FORTRAN IV, or for that matter any other language. An interesting but important benefit of this is that, we are able to have the documentation of the program even before we have the executable statements.

Modular programming means the program will be defined according to the flow of data and each module will represent a functional part of the system. Interfacing between modules is done only by passing of parameters, and each parameter is carefully defined to be either an input parameter, output

parameter, or input-output parameter so that modular independence can be achieved. Thus a change in the implementation of a module will not affect the coding of any other module in the system. The advantage of modular programming is to allow people to work on separate modules indepently, once the interfaces are defined.

3.2 The Data Unit

The basic input data to the system is a sine wave signal. (In the future, we plan to have actual sampled voice as our input signal.) Samples are taken from the signal and are passed through the system sample by sample. The sampling frequency, i.e. how often a sample is taken, is defined as a multiple of the Nyquist rate, and is input by the user. We analyzed and concluded that for our simulator, a sampling frequency of at least twice the Nyquist rate will be an acceptable approximation of the original signal. This is needed because of the integrations and differentiations that are done within the simulation. Also from the same study, a multiple of beyond 4 times the Nyquist rate seemed not to improve the results with any significance.

3.3 The Communication Scenarios

We proceeded to analyze how to implement the simulation of the various types and modes of calls possible within our system (refer to section 2.2) and concluded that it will be possible to study all the different possible combinations of types and modes of calls through the use of five communication scenarios. (We are able to combine different types of calls together if they have to go through the same basic modules of the simulator.) The five scenarios are:

1. Single Hop System:
For those calls which require the communication between two mobile units, without going through a gateway, we group them together and form scenario 1. In this scenario, a mobile will generate a call, for which the call will go up to the satellite, and be transponded down directly to the destination mobile. (figure 3.3.1)

2. Double Hop, Single Gateway System:
For those calls which require communication between two mobile units by going through a gateway, we group them together and form the scenario 2. In this scenario, a mobile will initiate a call, which will be transponded by the satellite to a

gateway. After processing by the gateway, the call is once again transponded by the satellite to the destination mobile. (figure 3.3.2)

3. Double Hop, Double Gateway System:

For those calls which have to go through two gateways, then it belongs to scenario 3. In this scenario, a mobile will generate a call, which is transponded to a gateway. The gateway forwards the call to the master control station. The master control station then routes the call to the destination gateway. Then the call is once again transponded by the satellite to the destination mobile. (figure 3.3.3)

4. Mobile-to-Wireline System:

For any call that was initiated by a mobile and was desired to go to a fixed phone or to a mobile serviced by the cellular system, we have scenario 4. In this scenario, the call is initiated by the mobile and then transponded by the satellite to a gateway. Then, the gateway forwards the call into the wireline telephone network. (figure 3.3.4)

5. Wireline-to-Mobile System:

If a call is initiated by a fixed phone or a mobile in a cellular system to a mobile in the LMSS, we resort to scenario 5. In this scenario, a call is received from the wireline

telephone network into a gateway. The call is then transponded by the satellite to the destination mobile. (figure 3.3.5)

Table 3.3.6 is a summary of the scenario selected based on the types of calls and modes of calls.

Table 3.3.6: Selection of Scenario

Class	Type	Scenario Number
1	1	1
	2	1
	3	1
	4	2
2	1	1
	2	1
	3	1
	4	2
3	1	1
	2	1
	3	1
	4	2
	5	2
	6	3
4	1	4
5	1	4
	2	4
	3	4
6	1	5
7	1	5
	2	5
	3	5

3.4 The Selection of the Scenario

When the simulator is executed, the user will first be asked for the type of call to be simulated. After the user had input his/her choice, then, based on the selection, the user will be asked to select the mode of call desired from those that are available for the selected type of call. Consequently, the scenario number is determined based on the user selection of the type and mode of call (refer to section 3.3).

Under the current version of the simulator, only the single hop system (scenario 1) is implemented. Any other scenario selection will not produce meaningful output at this time. The remainder of the scenarios are to be implemented during year two of this project.

First, tne type of calls that are available in the communication system is displayed as follows (refer to section 2.2):

```
Type of call available:
1:  M1->M2, rural mobile to rural mobile in same UHF-beam
2:  M1->M3, rural mobile to rural mobile in different UHF-beam,
            in same S-band
3:  M1->M4, rural mobile to rural mobile in different UHF-beam,
            in different S-band
4:  M1->FC1, rural mobile to fixed in same S-band beam
```

5: M1->FC2, rural mobile to fixed in different S-band beam

6: FC1->M1, fixed to rural mobile in same S-band beam

7: FC2->M1, fixed to rural mobile in different S-band beam

Then the user is asked to input a number from 1 to 7 which represent the type of call to be simulated.

After the user has input the type of call to be simulated, then the modes of call available within the type of call selected will be displayed (refer to section 2.3). For example if the user chose to simulate type 1, which is rural mobile calling rural mobile in the same UHF-beam and S-beam, then the mode of call will be displayed as follows:

MODE OF CALL AVAILABLE:

1: M1->M2, hard wired transponder

2: M1->M2, direct switched transponder

3: M1->M2, indirect switched transponder

4: M1->G1->M2, double hop system

Then the user is asked to input the mode of call to be simulated.

Based on the type of call and mode of call selected, the program will determine which scenario to be simulated (refer to section 3.3).

4. The Simulator

4.1 The Controllers

4.1.1 The Simulator Controller

The user can control the duration of the simulation run through three inputs: the duration of the input signal to be tested, the sampling frequency, and the range of SNR (signal to noise ratio) values to be tested.

The duration of the input signal to be tested and the sampling frequency will determine the number of samples to be generated and passed through the system for each SNR value to be tested (refer to section 4.1.3).

The number of different SNR value to be tested will then determine the number of iteration runs of the simulator required; one for each value to be tested.

Hence, the total time a simulation run will take is the product of the time it takes to process one sample, the sampling frequency used, the duration of the input signal to be tested, and the number of different SNR values to be tested.

4.1.2 The Master Controller

The purpose of the master controller is to drive the subcontroller to perform iterative runs of the simulator with the parameters as specified by the user.

First, it calculates the noise power for the given SNR of the current iteration and resets the simulation clock to 0. Then it calls the subcontroller to perform an iteration of the simulator (refer to section 4.1.3). Finally, it uses the master controller instrumentation package to measure the output SNR if requested by the user (refer to section 4.2.2).

The algorithm is as follow:

```
**************************************************************
/simulation master controller function/:
  <initialize simulation master controller function>
  DO <range of SNR>
     <calculate SNR>
     <calculate standard deviation for white noise>
     <perform simulation reset>
     <set flag for mobile xmtr>
     <perform a simulation run>
     <perform master controller instrumentation package>
 END <signal to noise ratio>
END /simulation master controller function/

**************************************************************
```

4.1.3 The Subcontroller

The purpose of the subcontroller is to perform an iteration of the simulator by passing samples of the original signal through the system.

This program utilizes a loop in which it first tests whether the current iteration has been completed by comparing the simulation clock against the duration of the input signals to be tested, as requested by the user. If the run is not finished, it calls the appropriate scenario (refer to section 3.3) to process the sample. Then, the subcontroller instrumentation package is called to accumulate or print the output signal as selected by the user (refer to section 4.2.1). The simulation clock is then updated by adding the sampling time interval to reflect the current simulation time. This cycle will be repeated until the proper number of samples have been processed. (refer to section 4.1.1).

The algorithm is as follow:

```
*****************************************************************
/simulation sub-control function/:
  DO UNTIL <end of current simulation run>
    DO /an iteration of the simulation run/:
      <perform the scenario>
      <update the simulation control function>
```

```
    END /an iteration of the simulation run/

   END

END /simulation sub-control function/

****************************************************************
```

4.2 The Instumentation

4.2.1 The Instrumentation Package

Under the current version, the user has the choice between two types of output: to compare the output signal with the input signal (graphic and tablular plots are provided) (refer to section 4.2.3), or to compare the output SNR against the input SNR (refer to section 4.2.2).

The output from the simulator can be used to study the following:

a) Study effect of interference on FM;

b) Study effect of modulation and fading on frequency spectrum;

c) Effect of use of filter on SNR;

d) Effect of channel on SNR;

e) Effect of use of space diversity receiver;

f) All combinations of the above.

4.2.2 The Master Controller Instrumentation Package

The purpose of this instrumentation package is to calculate the output SNR and compare it against the input SNR, if requested by the user.

During an iteration, the subcontroller instrumentation package measures the output signal power and the output noise power (refer to section 4.2.3). Based on this data, the master controller instrumentation package calculates the output SNR by:

10 * log (output signal power / output noise power)

and prints a table of the input SNR's tested along with the output SNR.

The algorithm is as follow:

```
*************************************************************
/simulation master controller instrumentation package/:
           <compute signal to noise ratio>
           <display signal to noise ratio>
           <reset output power>
END <simulation master controller instrumentation package>
*************************************************************
```

4.2.3 The Subcontroller Instrumentation Package

The purpose of this instrumentation package is to print the output signal along with the input signal, or to measure the output signal power and noise power.

Each time when a sample is taken from the original signal, two copies of the sample are made and sent through the system. One sample will pass through the entire communication system without accumulating any noise. This sample is called the signal without noise. The other sample will also be passed throught the system but will be affected by noise and interference. This sample is called the signal with noise.

When the user requests to print the output signal, both the sample with noise and sample without noise will be printed (and plotted) together with the original sample. In this way, the effect of the communication system and the effect of noise and interference can be studied by comparison on a sample to sample basis.

When the user requests to measure the output SNR as a function of the input SNR, then the signal power and noise power are measured in this instrumentation package (refer to section 4.2.2).

The power of the signal is calculated as the sum of the square of the output signal without noise. The power of the noise is calculated as the sum of the square of the output noise which in turn is calculated by subtracting the signal component from the output signal with noise.

The algorithm is as follow:

```
*************************************************************
<simulation subcontroller instrumentation package>
  CASE <type of measurement>
    /1/: <display original signal & output signal>
    /2/: <accumulate the output signal power and output noise
          power>
  END
END /simulation subcontroller instrumentation package>
*************************************************************
```

User's Guide

This user's guide will provide the user with a step by step explanation of the questions that will be asked during a session with the simulator. References are given in each question so that the user can refer to the appropriate sections if an in depth discussion about the topic is desired.

The first question asked will be whether if the session is an online or batch session. The user should answer a 'Y' if it is an online session, or a 'N' if it is a batch session.

First, the type of calls that are available in the communication system is displayed as follows (refer to section 2.2):

Type of call available:

1: M1->M2, rural mobile to rural mobile in same UHF-beam

2: M1->M3, rural mobile to rural mobile in different UHF-beam, in same S-band

3: M1->M4, rural mobile to rural mobile in different UHF-beam, in different S-band

4: M1->FC1, rural mobile to fixed in same S-band beam

5: M1->FC2, rural mobile to fixed in different S-band beam

6: FC1->M1, fixed to rural mobile in same S-band beam

7: FC2->M1, fixed to rural mobile in different S-band beam

Then the user is asked to input a number from 1 to 7 which represent the type of call to be simulated.

After the user has input the type of call to be simulated, then the modes of call available within the type of call selected will be displayed (refer to section 2.3). For example if the user chose to simulate type 1, which is rural mobile calling rural mobile in the same UHF-beam and S-beam, then the mode of call will be displayed as follows:

MODE OF CALL AVAILABLE:

1: M1->M2, hard wired transponder

2: M1->M2, direct switched transponder

3: M1->M2, indirect switched transponder

4: M1->G1->M2, double hop system

Then the user is asked to input the mode of call to be simulated.

Based on the type of call and mode of call selected, the program will determine which scenario to be simulated (refer to section 3.3). The questions asked from this point on will vary depanding on the features to be simulated within the scenario. Under the current version, only the single hop system (scenario 1) is implemented, so we shall focus our discussion on those

questions that will be asked when the single hop system is to be simulated. Then the user will be asked to input the frequency of the baseband signal in hertz. The baseband signal has an upper limit of 3000 Hz (refer to section 5).

Then the user will be asked to input the power of the baseband signal in watts. The power of the baseband signal has to be less than or equal to 0.5 watts, so that the maximum instantaneous frequency deviation is less than or equal to 12000 Hz (refer to section 5).

Then the user will be asked to input the carrier power in watts (refer to section 5).

Then the user may be asked to input the frequency deviation. Under the current version, the frequency deviation is fixed at 12,000 Hz in order for the simulator to meet AMPS specifications (refer to section 5).

Then the user will be asked to input the sampling frequency in terms of a multiple of the Nyquist rate. Acceptable values are from twice the Nyquist rate to four times the Nyquist rate. Twice the Nyquist rate is the minimum in order for the simulator to integrate and differentiate currectly. Four times the Nyquist rate is the maximum since any increase of the sampling frequency

beyond this point will not improve any signigicant approximation, but will prolong the run time substantially (refer to section 3.2).

Then the user may be asked to input the carrier frequency. Under the current version of the simulator, the carrier frequency is not used in the simulation (refer to section 5).

Then the user will be asked whether compressor/expandor (AMPS specs) is used in the transmitter and receiver (refer to section 5).

Then the user will be asked whether pre-emphasis and de-emphasis filters (AMPS specs) are used in the transmitter and receiver (refer to section 5).

Then the user will be asked whether fading is present in the uplink channel and whether the interference source will be faded. If fading is present, the the user will be asked to choose the type of fading channel,(refer to section 6) as following:

1. No specular component (Raleigh fading)
2. Specular component, but shortest path
3. Specular component, but mean path

The the characteristics of the fading channel will be asked as follow: the multipath spread time in micro-second; the doppler

spread bandwidth in Hertz; and the specular-to- multipath power ratio in Db. If fading is not present, then these questions will not be asked.

Then the user will be asked whether if fading is present in the downlink channel. The same types of questions will be as forked as for the uplink channel which was discussed in the previous paragraph.

Then the user will be asked whether multiple SNR values are to be tested. If it is, then the user will be asked to input the ranges of SNR's for different devices by specifying the initial value, the increment value, and the ending value of the SNR's. If multiple SNR value is not to be tested, then the user will be asked to input a SNR value for each device. Under the current version of the simulator, only the Gaussian noise in the mobile receiver is implemented (refer to section 4.1, 4.2).

Then the user is asked whether if space diversity receiver is used. If it is present, then the user will be asked to input the duration of the decision period which is the duration of lapse time before a decision is made to determine which receiver is receiving a stronger signal.

Then the user is asked to input the approximate duration of

the input signal to be tested in seconds. There is a boundary imposed as from 0.01 second to 9.99 second (refer to section 4.1).

Finally, the user is asked to input the type of performance to be measured. Under the current version, the user has the choice of choosing either to compare the recovered output signal to original input signal, or to measure the output signal to noise ratio (refer to section 4.2).

5. The Transmitter

5.1 FM Generation

An FM signal is generated in this stage. A frequency modulated signal can be generated using the device represented by the block diagram shown in Fig 5.1.1.

We have :

$$v(t) = \sqrt{2P_c}\cos\left(\omega_c t + 2\pi\Delta f \int_{-\infty}^{t} m(\tau)\,d\tau\right) \qquad (5.1.1)$$

Where

v(t) = output of device represented in fig. 5.1.1

Pc = power of the carrier

Wc = carrier angular frequency

t = time

Δf = frquency deviation

m(τ) = modulating signal

m($-\infty$) = 0

$$\frac{d}{dt}\left(\omega_c t + 2\pi\Delta f \int_{-\infty}^{t} m(\tau)\,d\tau\right) = \omega_c + 2\pi\Delta f\, m(t) \qquad (5.1.2)$$

Since the deviation of the instantaneous angular frequency is directly proportional to the modulating signal m(t) , the combination of integrator and phase modulator constitutes a device for generating a frequency modulated signal.

The modulating signal m(t) or "intelligence" is obtained from

the signal source generator. Before feeding it to the FM modulator, some optional signal processing is done on the modulating signal namely: preemphasis and compression.

The output of the FM modulator could be passed through a nonlinear stage and a filter before sending it to the satellite, if so is desired. (at this point in time , neither the nonlinearity nor the filter are implemented). A block diagram of the transmitter is shown in Fig. 5.1.2.

The algorithm is as follows:

```
/ Mobile to satellite transmitter/:
IF<mobile to satellite transmitter flag is on>
THEN <generate a new sample>
     <process the signal>
     <generate the FM signal>
     <pass the FM signal through a non-linearity>
     <filter the signal>
     <send the signal to the satellite>
ELSE <send unmodulated signal>
END/Mobile to satellite transmitter/
```

5.2 The Signal Source Generator

Samples of the modulating signal are generated in this stage In the present version of the simulator the modulating signal m(t) is a cosine.

$$m(t)=\sqrt{2P_s}\cos(2\pi F_s t) \qquad (5.2.1)$$

Where:

Ps = power of the modulating signal

Fs = frequency of the modulating signal

t = time

The algorithm is as follows:

/Signal source generator routine/:

<sample of the modulated signal is calculated>

END/Signal source generator routine/

5.3 Transmitter Signal Processor

The modulator is preceded by the following four voice-processing stages:

a.- Compressor

b.- Preemphasis

c.- Deviation limiter

d.- Post deviation-limiter filter

A block diagram of the transmitter signal processor is shown in Fig. 5.3.1.

5.3.1 The Compressor

Basically, a compressor is a nonlinear device which reduces the amplitude range of an input signal; so that it falls whithin a predetermined range. At the receiver the inverse operation, expanding, is performed; so that the signal is restored to its appropiate range. The combined operation of compressing and expanding is called companding. Companding is used in telephone systems to reduce nonlinear distortion and to compensate for signal level difference between loud and soft talkers.

The compressor simulated is a 2:1 syllabic compandor. For every 2 db. change in input level, to this compressor, the change in output level is 1 db.

The algorithm is as follows:

/ The compressor/:

IF <compressor flag is off>

THEN < end >

ELSE < compress the sampled signal >

END/ The compressor/

5.3.2 The Preemphasis Filter

In an FM system a source of noise improvement is provided by preemphasis of the high frequencies at the transmitter and corresponding deemphasis at the receiver. This improvement is due to the adequate choice of the filters to minimize noise. The power spectral density of the noise at the output of the demodulator increases proportionally with the square of the frequency. So it is possible to choose a filter such that its frequency response falls off as the frequency increases.Therefore, noise will be reduced; and so will the signal. But if the signal is filtered in the transmitter in such a way that the attenuation of the filter in the receiver is cancelled,then the original signal is recovered in the receiver [5.1].

The preemphasis filter Fig. 5.3.2 raises the audio level at a rate of 6 db/octave above 300 Hz. and the deemphasis filter Fig 5.4.3 in the receiver decreases the audio output at 6db./octave; thus producing a flat output audio response. However, the noise passes only through the deemphasis filter and noise is thereby suppressed to some extent.

The simulated preemphasis filter was designed using the inverted bilinear transformation technique from a one pole analog filter. The implementation corresponds to direct form II [5.2]

shown in Fig. 5.3.3.

The algorithm is as follows:

```
/Preemphasis filter routine/:
 IF <preemphasis flag is on>
    THEN <feed the sampled input to the filter>
    ELSE < end>
    end
END/Preemphasis filter routine/
```

5.3.3 The Deviation Limiter

According to the working paper entitled Cellular System Mobile Station - Land Station Compatibility Specification, January 1980 prepared by an EIA Ad Hoc Comittee to address the question of technical compatibility for cellular service; "For audio inputs applied to the transmitter signal processing stage, a mobile station must limit the instantaneous frequency deviation to + 12 KHz. ". In the simulator this specification is satisfied when the user enters the power of the modulating signal (which has to be less than 0.5 Watts).

5.3.4 The Post Deviation-Limiter Filter

This filter is a low pass filter whose attenuation characteristic is given by :

$$40 \text{ Log } (f/3000) \text{ db.} \qquad f > 3000 \text{ Hz.}$$

The purpose of this filter is to bardlimit the modulating signal; so that the bandwidth allocated to the FM signal is not exceeded.

The bilinear transformation technique is used to design this filter from a 2-pole analog filter. The result is shown in Fig. 5.3.4.

5.3.5 The Modulator

As explained in 5.1 an FM signal is generated by passing the modulating signal through an integrator and a phase modulator. The output of the phase modulator is given by:

$$v(t) = \sqrt{2Pc}\ \mathrm{Cos}\,(\,Wc\ t + \phi(t)\,) \qquad (5.3.1)$$

where:

$$\phi(t) = 2\pi\,\Delta f \int_{-\infty}^{t} m(\tau)\,d\tau \qquad (5.3.2)$$

Therefore, one way to generate v(t) would be to add $\phi(t)$ to the term Wct , and then take the cosine of that sum. However, this would mean that the sampling frequency required in the simulator be at least twice as much as Fc. Since Fc is a very high frequency, the amount of computer time necessary to process a few seconds of the modulating signal will be astronomical. To avoid this problem, the modulator only generates and transmits the inphase and quadrature components of v(t), namely $\sqrt{2Pc}\ \mathrm{Cos}\,\phi(t)$ and $\sqrt{2Pc}\ \mathrm{Sin}\,\phi(t)$.

$$v(t) = \sqrt{2P_t}\left[\cos w_c t \cos\phi(t) - \sin W_c t \sin\phi(t)\right] \qquad (5.3.3)$$

a phasor representation of v(t) is shown in Fig. 5.3.5.

To show that such an approach will still represent the simulation of an FM waveform consider that noise

$$n(t) = n_c(t)\cos w_c t - n_s(t)\sin w_c t \qquad (5.3.4)$$

is added to the transmitted signal. Then the received signal is

$$\Omega(t) = \sqrt{2P_c}\left[\left(\cos\phi + \frac{n_c(t)}{\sqrt{2P_c}}\right)\cos w_c t - \left(\sin\phi + \frac{n_s(t)}{\sqrt{2P_c}}\right)\sin w_c t\right] \qquad (5.3.5)$$

and in terms of envelope R(t) and phase $\alpha(t)$, we have

$$\Omega(t) = \sqrt{2P_c}\, R(t)\cos(w_c t + \alpha(t)) \qquad (5.3.6)$$

where

$$R(t) = \left[\left(\cos\phi + \frac{n_c(t)}{\sqrt{2P_s}}\right)^2 + \left(\sin\phi + \frac{n_s(t)}{\sqrt{2P_s}}\right)^2\right]^{1/2} \qquad (5.3.7)$$

and

$$\alpha(t) = \tan^{-1}\left[\frac{\sin\phi + n_s(t)/\sqrt{2P_s}}{\cos\phi + n_c(t)/\sqrt{2P_s}}\right] \qquad (5.3.8)$$

When there is no noise $\alpha(t) = \phi(t)$ as expected.

A basic FM receiver, to recover the modulating signal from v(t) or r(t) consists of an IF filter, a limiter, a discriminator, and a baseband filter as shown in Fig. 5.3.6.

If v(t) is the input to this receiver, the signal at the output of the limiter is:

$$V_L(t) = A_L \operatorname{Cos}(Wct + \alpha(t)) \qquad (5.3.9)$$

Where A_L is the amplitude of the limiter output. Assuming $A_L = 1$

the output of the differentiator is

$$v_d(t) = -\left(\omega_c + \frac{d\alpha(t)}{dt}\right)\sin(\omega_c t + \alpha(t)) \qquad (5.3.10)$$

Finally the output of the baseband filter is

$$v_o(t) = \frac{d}{dt}\alpha(t) \qquad (5.3.11)$$

where we have assumed that the baseband filter rejects dc.

The demodulated output is then

$$v_o(t) = \frac{d}{dt}\left[\tan^{-1}\frac{\sin\phi + n_s/\sqrt{2P_c}}{\cos\phi + n_c/\sqrt{2P_c}}\right] \qquad (5.3.12)$$

and if there is no noise

$$v_o(t) = 2\pi\Delta f\, m(t) \qquad (5.3.13)$$

as expected.

The simulated FM modulator consists of an integrator, an amplifier with a gain $2\pi\Delta f$, and the stage that generates the inphase and quadrature components.

The algorithm is the following:

/FM modulator/:

< integrate the modulating signal >

< multiply by $2\pi\Delta f$ >

< generate the inphase and quadrature components >

END/FM modulator/

5.3.6 Non-linearity Stage

In the current version of the simulator this stage is only a

'dummy' subroutine. The signal is currently passed through a stage of gain 1 . If a nonlinearity is required, it will be easily implemented since the skeleton of the subroutine has alredy been designed.

5.3.7 Filter

If additional filtering of the modulated signal is required, this subroutine could provide such filtering. The current version of the simulator has a "dummy" subroutine instead of a filter. Again, the point should be made that a filter could be easily implemented since the skeleton of the subroutine has already been designed.

5.4 The Receiver

In the FM simulator, as it was stated before, the transmitted signals are the inphase component($\sqrt{2P}\,\cos\phi(t)$) , and the quadrature component ($\sqrt{2P}\,\sin\phi(t)$). The Fm receiver used to recover the modulating signal is shown in Fig. 5.4.

Comparing the simulated receiver with the standard FM receiver of Fig. 5.3.6, notice that both have filters at the receiver input to limit the incoming noise. There is a limiter in the conventional FM receiver while in the simulator, we do not need one.(simulator was designed in such a way that an ideal limiter was assumed). While the conventional FM receiver has a discriminator (differentiator and an envelope detector); the simulator only has a differentiator. The envelope detector is not needed since it is a baseband simulation. Finally, both receivers have baseband filters.

The algorithm is as follows:

/Mobile satellite receiver/:

< using space diversity, the strongest signal is selected >

< demodulation is performed >

< process the demodulated signal >

END/Mobile satellite receiver/

5.4.1 Diversity

If the signal perturbation in the channel were only due to additive gaussian noise, the basic receiver shown in Fig 5.3.6

would suffice to recover the modulating signal. However, non-gaussian channel perturbations are very common in practice. For example, multiple transmission paths due to stratifications in the transmission medium or reflecting objects is one of them. One way to combat this kind of degradation (fading) on the signal is by using a space diversity receiver. If dual diversity is used, such receiver consists of two antennas, two IF filters, a signal selector, a discriminator, a baseband filter, and a processing stage. The two antennas are spaced sufficiently apart; so that the received signals are independent of one another. The IF filters limit the noise going into the receiver while the signal selector picks the strongest signal.

5.4.2 The Space Diversity Receiver

Two antennae will be used to receive the signal for the mobile receiver. The antennae will be placed sufficiently apart so that the noise effect and the fading channel effect will be statistically independent at each receiver. Thus, one antenna may be receiving a stronger signal while the other is receiving a weaker signal. One of these received signals will be passed through to the mobile receiver for demodulation based on the following selection algorithm:

A decision period is that duration of time between which a decision is made to determine which antenna is receiving a stronger signal. The duration of the decision period is an user

input variable.

During a decision period, the power of the signals received at both antennae is accumulated using an envelope detector (integrate and dump technique). At at the end of a decision period, the power of the signals received at both antennae are compared to determine which antenna had been receiving stronger signals. Whichever one is receiving a stronger signal will be chosen to pass through its received signals to the receiver for demodulation during the next decision period.

The algorithm is as follow:

```
*************************************************************
/Space diversity receiver routine/:
  IF <space diversity receiver is on> THEN BEGIN
    <accumulate signal power>
    <increment time counter>
    IF <decision time> THEN BEGIN
     <reset time counter>
      IF <receiver (1) is stronger than receiver (2)>
        THEN <choose receiver (1)>
        ELSE <choose receiver (2)>
      <reset accumulation signal power>
                          END
                                         END
  <choose the output signal>
END /Space diversity receiver routine/
```

****

5.4.3 The FM demodulator

The inputs to the simulated FM demodulator are the inphase and quadrature components. They could be faded, perturbed with gaussian noise, and/or a cochannel interference. In other words any possible combination could be handled by the simulator. The simulated FM demodulator is shown in Fig. 5.4.

In this demodulator the blocks which deserve some comment are the differentiator and the baseband filter since the other blocks are self-explanatory.

The differentiation is implemented by taking the backward difference of the arc tangent of the ratio of quadrature and phase components. The difference is divided by the time between two consecutive samples to obtain the derivative.

The baseband filter will supress the noise above 3 KHz.,therefore the signal to noise ratio will improve. This filter has been designed using the bilinear transformation technique from a 2 pole analog filter.

The algorithm is as follows:

/The FM demodulator/:

< obtain ratio of inphase and quadrature components >

< obtain Arc tan of such ratio >

< perform differentiation >

< filter the demodulated signal >

END/The FM demodulator/

5.4.4 Receiver Signal Processor Stage

The purpose of this stage is to undo whatever processing was done in the transmitter. Since a compressor and a preemphasis filter are available in the transmitter; an expander and a deemphasis filter have been designed into this stage.

The simulated expandor is the portion of a 2:1 syllabic compandor. For every 1 db. change in input level to the expandor, the change in output level is 2 db.

The deemphasis filter has a frequency response ; such that it cancels out the preemphasis done in the transmitter. The deemphasis characteristic has a -6 db/octave response between 300 Hz. and 3000 Hz.

The algorithm is as follows:

```
/Receiver Signal Processor/:
  IF < deemphasis flag is on>
     THEN < do deemphasis >
  IF < expandor flag is on >
     THEN < do expansion >
END/Receiver signal processor/
```

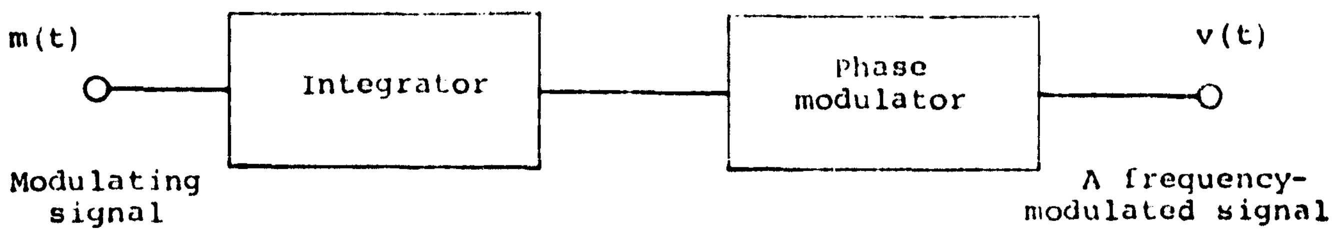


Fig. 5.1.1 FM Generation

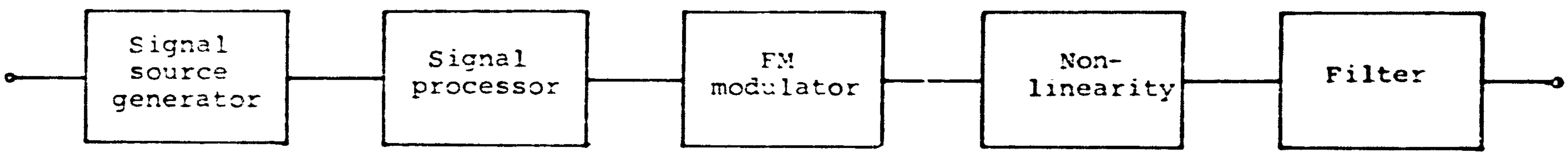


Fig 5.1.2 Transmitter

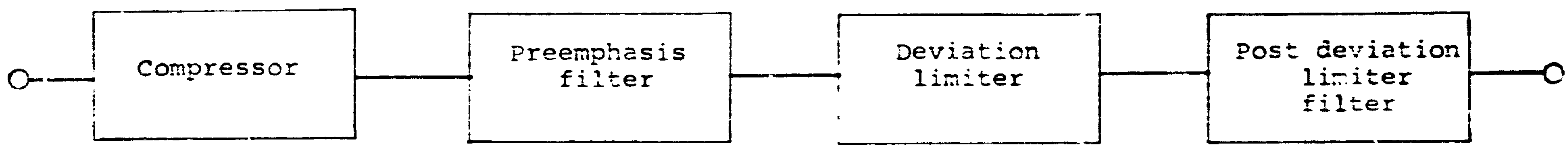
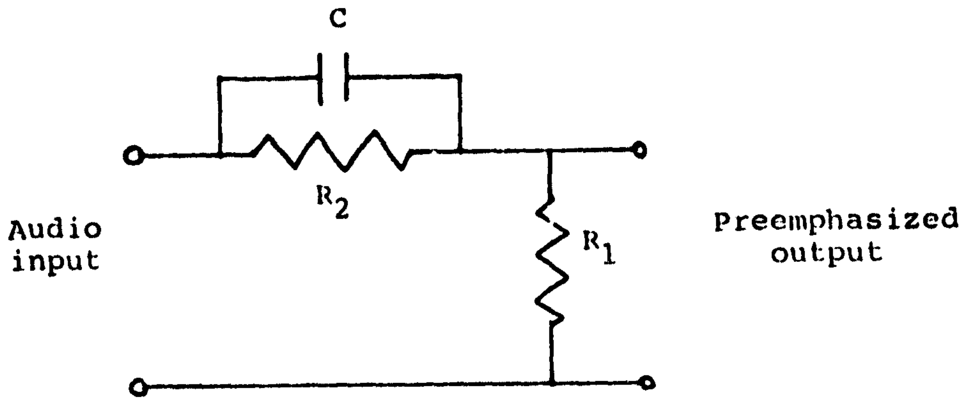
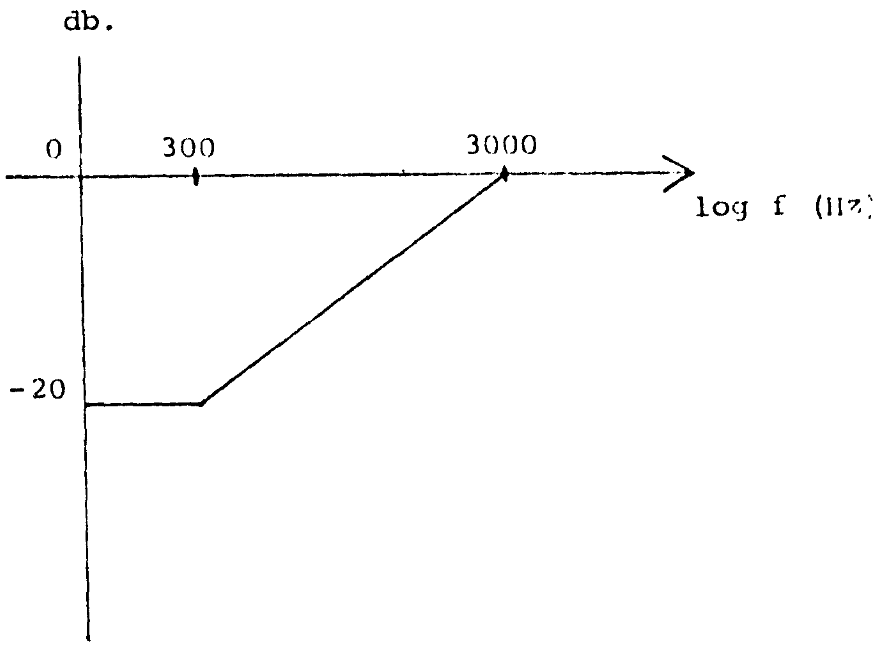


Fig 5.3.1 Transmitter Signal Processor



(a) Circuit



(b) Frequency Response

Fig. 5.3.2 Preemphasis Filter

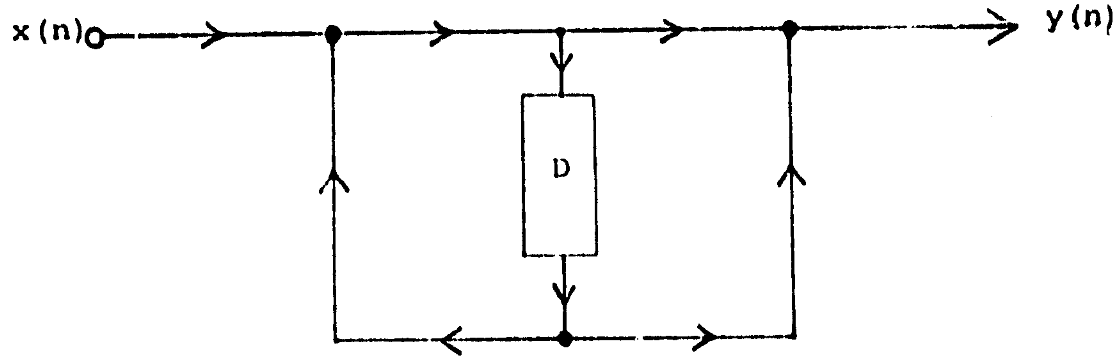


Fig 5.3.3 Preemphasis Filter

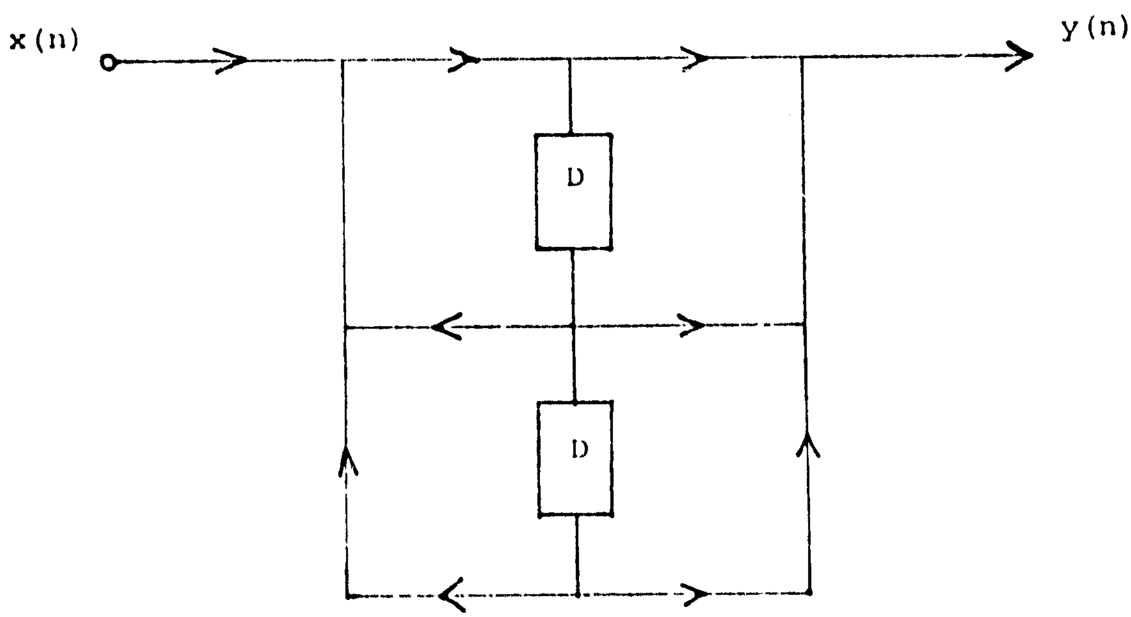


Fig 5.3.4 The Post Deviation-Limiter Filter

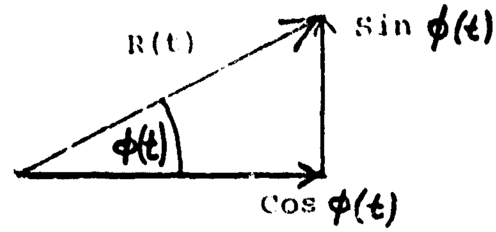


Fig 5.3.5 FM Quadrature Components

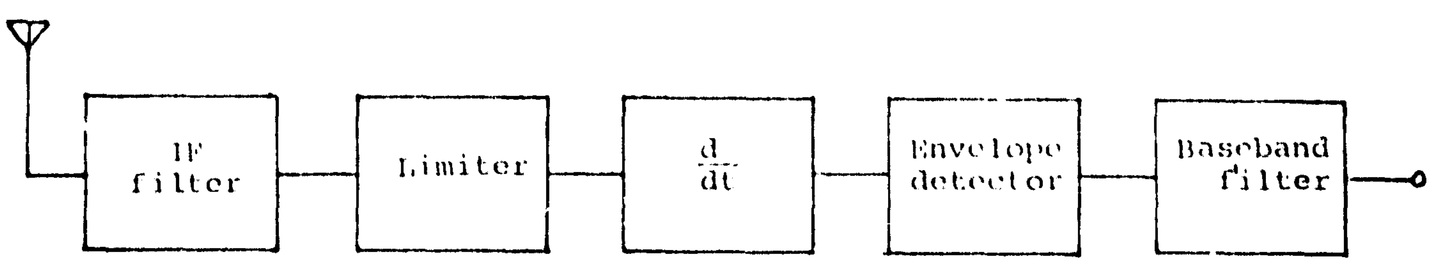


Fig 5.3.6 FM Receiver

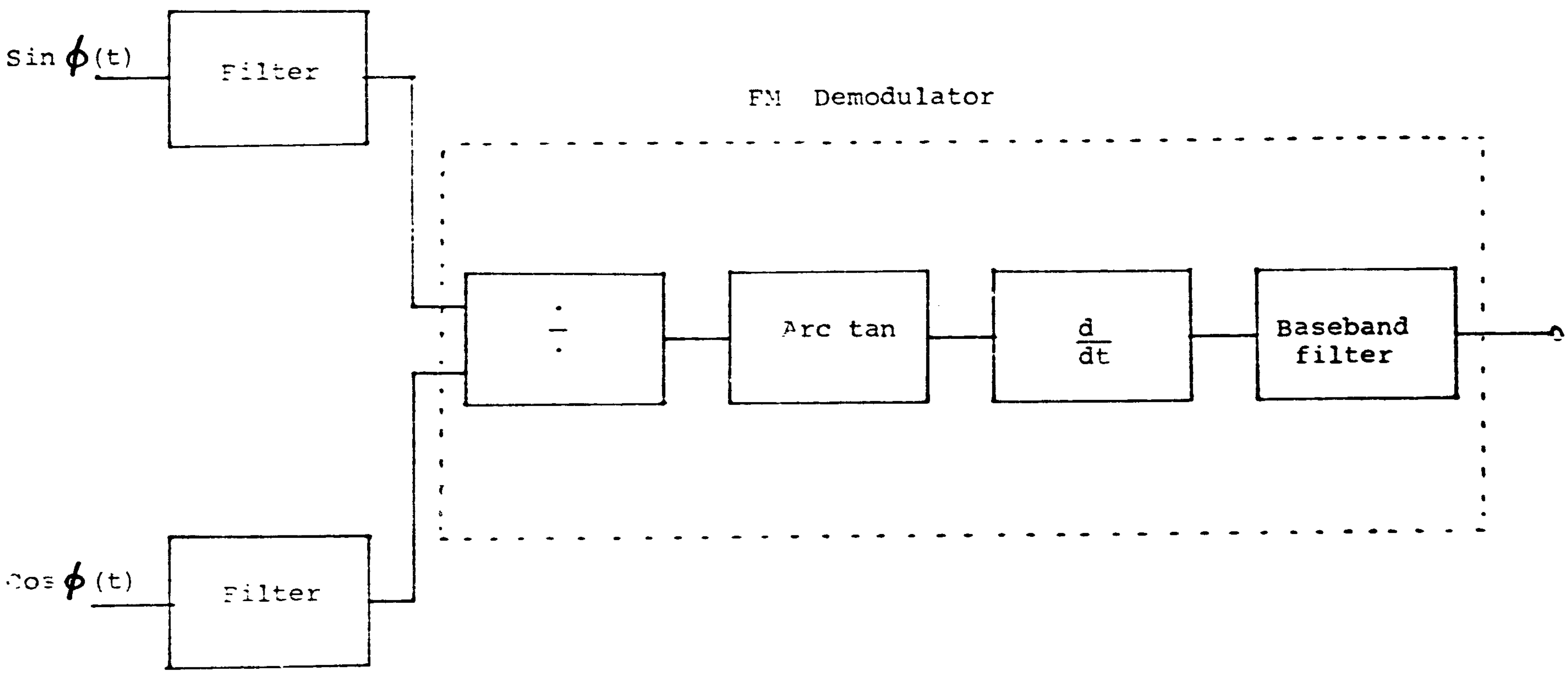
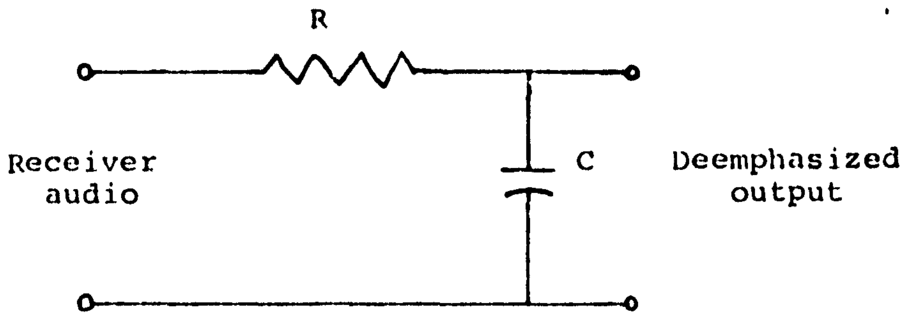
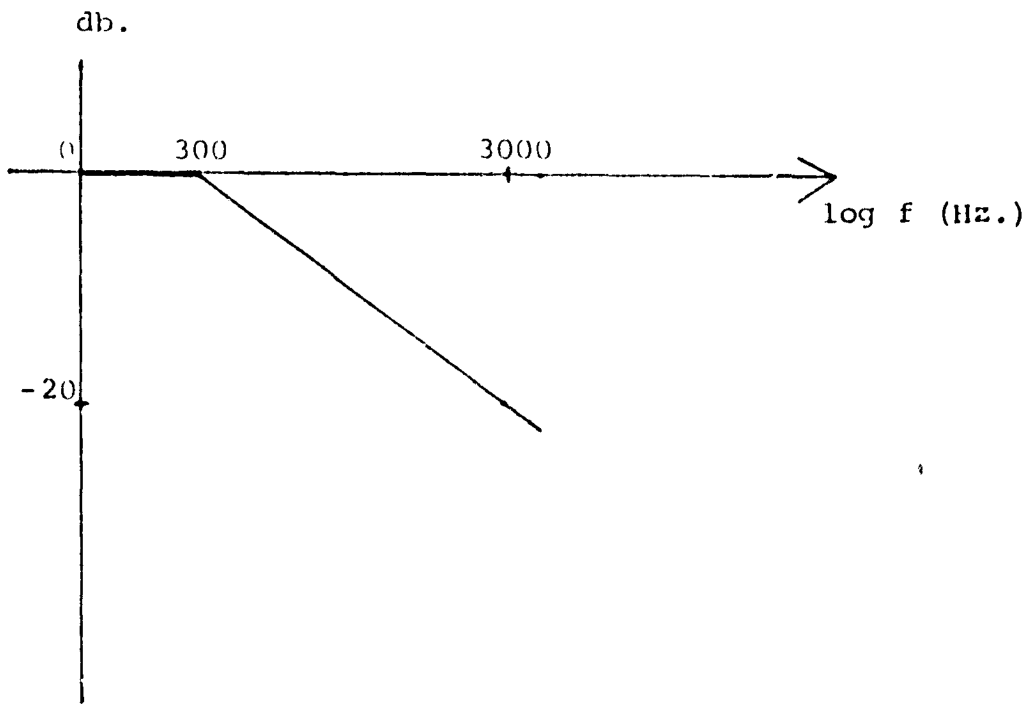


Fig 5.4 Simulated FM Receiver



(a) Circuit



(b) Frequency Response

Fig. 5.4.3 Deemphasis Filter

6.1. The Fading Channel

A frequently used approach to long distance communications is via satellite channels, as in the case of this simulation. Frequently, this radio transmission is via a channel which exhibits both time and/or frequency dependent variations in signal amplitude and phase. Such a channel is refferred to as a fading channel.

Fading of radio signals can occur in a variety of ways. The most common cause of fading is due to multipath. In multipath fading, the transmitted signal arrives at the receiver via more than one propagation path, and frequently with time and/or frequency varying relative phase differences. However, at the receiver, these paths are indistinguishable, and what is really received is the resultant of these paths. Consequently, the resultant received waveform is a replica of the transmitted signal modified by varying amplitude and phase components. When the number of components of the multipath are large, the resultant will have Rayleigh statistics; thus being called Rayleigh fading.

It is also possible, that either due to a free line of sight path, or a strong reflection, that one or more of the components will be much stronger than the rest, thus carrying most of the signal power. These components are called specular components. Whenever the channel output consists of a strong stable specular signal plus a faded version of this signal, the resultant will have Rician statistics, thus being called Rician fading. This type of transmission medium is much more preferable than one which

exhibits Rayleigh fading, since a significant amount of the power is contained in a major stable communication path, thus reducing the relative power of the randomly varying multipath components. The fading channel is modeled as a wide-sense-stationary uncorrelated scattering (WSSUS) channel. Such a model is general enough to exhibit both time and/or frequency selectivity and imposes no restriction on the fading rate. A commonly used model of a fading channel is the tap-delay-line model (Fig. 6.1.1)-a model which is employed in this simulation.

This model takes the transmitted signal, which is of the form

$$S(t)=\sqrt{2P_c}\,u(t)\cos(\omega_c t+\theta(t)+\theta_o) \qquad (6.1.1)$$

where

$u(t)$ = envelope of the signal, with $E\{u^2(t)\}=1$,
P_c = power of the carrier,
$\theta(t)$ = instantaneous phase of the carrier,
ω_c = carrier angular frequency, and
θ_o = arbitrary phase shift;

and operates on it, producing the received signal

$$r(t)=\sqrt{2P_c}\sum_{n=0} R(t-nT)u(t-nT)\cos(\omega_c t+\theta(t-nT)+\phi(t-nT)+\theta_o) \qquad (6.1.2)$$

where

T = delay per tap (delay between the multipath components)

$R(t)$= Rayleigh distributed transmission (tap) gain(s)

$\phi(t)$= uniformly distributed $(0,2\pi)$ phase.

The quadrature component representation of thermal noise

$$n(t)= n_c(t)\cos(\omega_c t+\theta_o)+n_s(t)\sin(\omega_c t+\theta_o) \qquad (6.1.3)$$

where $n_c(t)$ and $n_s(t)$ are uncorrelated Guassian random processes of zero mean value and of equal variance, is then added to the signal.

For convenience, the following notation will be introduced. Let

$$g(t) = R(t)\cos\phi(t)$$

and

$$h(t) = R(t)\sin\phi(t). \tag{6.1.4}$$

It can be shown that both g(t) and h(t) are not only Guassian but are also statistically independent of each other and have zero means and equal variances. The resultant signal (the sum of (6.1.2) and (6.1.3)) in terms of quadrature components is then found to be

$$\begin{aligned} r(t) = & \sum_{n=0} \sqrt{2P_c}\, g(t-nT)u(t-nT)\cos(\omega_c t+\theta_o+\theta(t-nT)) \\ & - \sum_{n=0} \sqrt{2P_c}\, h(t-nT)u(t-nT)\sin(\omega_c t+\theta_o+\theta(t-nT)) \\ & + n_c(t)\cos(\omega_c t+\theta_o) - n_s(t)\sin(\omega_c t+\theta_o) \end{aligned} \tag{6.1.5}$$

Expanding further and including the specular component, it can be seen that

$$\begin{aligned} r(t) = & \sqrt{2P_c}\Big[u(t)\cos\theta(t) + \sum_{n=1} u(t-nT)\Big\{g(t-nT)\cos\theta(t-nT) \\ & \quad - h(t-nT)\sin\theta(t-nT)\Big\} + n_c(t)\Big]\cos(\omega_c t+\theta_o) \\ & - \sqrt{2P_c}\Big[u(t)\sin\theta(t) + \sum_{n=1} u(t-nT)\Big\{g(t-nT)\sin\theta(t-nT) \\ & \quad + h(t-nT)\cos\theta(t-nT)\Big\} + n_s(t)\Big]\sin(\omega_c t+\theta_o) \end{aligned} \tag{6.1.6}$$

This expression is simulated as shown in Fig. 6.1.3 as well as in the present simulation.

In the case of FM (see report 5.1) we have

$$u(t) = 1, \tag{6.1.7}$$

and

$$\theta(t) = 2\pi\Delta f \int_{-\infty}^{t} m_s(\tau)\,d\tau = \phi_s(t) \tag{6.1.8}$$

where

$m_s(t)$ = modulating signal

Δf = frequency deviation.

Thus the received signal for FM reduces to

$$\begin{aligned} r_{FM}(t) = \sqrt{2P_c}\Big[&\cos\phi_s(t) + \sum_{n=1}\Big\{g(t-nT)\cos\phi_s(t-nT) - h(t-nT)\sin\phi_s(t-nT)\Big\} \\ &+ n_c(t)\Big]\cos(\omega_c t + \theta_o) \\ - \sqrt{2P_c}\Big[&\sin\phi_s(t) + \sum_{n=1}\Big\{g(t-nT)\sin\phi_s(t-nT) + h(t-nT)\cos\phi_s(t-nT)\Big\} \\ &+ n_s(t)\Big]\sin(\omega_c t + \theta_o). \end{aligned} \tag{6.1.9}$$

This expression is simulated as shown in Fig. 6.1.2, and is what presently occurs in the overall simulation in the absence of interference. It should be noted that if interference is present, and if it undergoes fading, it is faded independently of the signal, and added at the end.

It was shown (sec. 5.3.5) that the output of the FM demodulator is

$$v_o(t) = \frac{d\alpha(t)}{dt} \tag{6.1.10}$$

where

$$\alpha(t) = \text{ARCTAN}\ \frac{\text{quadrature-phase component}}{\text{in-phase component}} \quad . \tag{6.1.11}$$

From equation (6.1.9),(6.1.10) and (6.1.11), we thus have that

$$v_o(t) = -\frac{d}{dt}\tan^{-1}\frac{\left[-\sin\phi_s(t) + \sum_{n=1}\left\{g(t-nT)\sin\phi_s(t-nT) + h(t-nT)\cos\phi_s(t-nT)\right\} + n_s(t)\right]}{\left[\cos\phi_s(t) + \sum_{n=1}\left\{g(t-nT)\cos\phi_s(t-nT) - h(t-nT)\sin\phi_s(t-nT)\right\} + n_c(t)\right]}. \tag{6.1.12}$$

We note that in the absence of fading and noise, our received output is

$$v_o(t) = \frac{d}{dt}\phi_s(t) = 2\pi\Delta f\, m_s(t) \tag{6.1.13}$$

which is our original signal. It is also important to note that as the power of the multipath path coefficients increase (i.e. specular-to-multipath power ratio decreases), the effect of fading on the output signal-to-noise ratio becomes significant, even when the input signal-to-noise ratio is large. In an analagous sense, when the power of the multipath components is small compared to the power of the noise terms, the effect of the input SNR value becomes the predominant factor in determining the output SNR value.

This phenomena can be seen on a number of sample simulation runs included within the report. In the first run (Fig. 6.1.4), we have a specular-to-multipath power ratio of 100 db, with a doppler spread bandwidth of 1 Hz, and a multipath spread time of 500 microseconds as typical fading channel parameters. We see here, as predicted above, that since the multipath components contain practically no power, the output SNR values are for all practical purposes a function of the input SNR values. Note, that

if these are compared to a simulation run without fading present (Fig. 6.1.5), the results are practically the same. In Fig 6.1.6, the specular-to-multipath power ratio is 35 db with all other parameters remaining unchanged. In this case the effects of fading are still very small, but can now be seen. We note that there is a slight flattening out in the output SNR values; a drop in the output SNR values of about 3 db as compared with the previous case. We next take a look at what happens with a specular-to-multipath power ratio of 20 db (Fig. 6.1.7), with all other parameters remaining the same. Here we already note a significant flattening out of the output vs input SNR values. As a last example, we look at a case (Fig. 6.1.8) where the specular-to-multipath power ratio is 10 db. In this case, the flattening out of the output vs input SNR curve is very pronounced, as projected by the above analysis.

The program also has the facility of a space diversity receiver. It is found that if such a receiver is used, the output results can be improved upon.

There are four questions which are asked by the simulation in order to set up the fading channel simulator. The first, gives the user the different possible fading scenarios available, namely:

(1) No specular component (Rayleigh Fading),

(2) Specular component (but shortest path), and

(3) Specular component (but mean path).

The second question prompts the users for the total multipath spread time. This is used to calculate the number of taps in the

delay line using the algorithm:

$$NTAPS = FMB * TSPRD + 1 , \qquad (6.1.14)$$

where

NTAPS = number of taps,

FMB = FM bandwidth, and

TSPRD = total multipath spread time.

From this the total number of samples in the delay line is calculated, using

$$TDLSMP = (NDLY * IFIX(1./TS*FMB))/2 + 1 \qquad (6.1.15)$$

where

TDLSMP = total delay line samples,

NDLY = NTAPS - 1 (number of delay elements), and

TS = sample time.

Then the doppler spread bandwidth is also asked, which is used to set up the 1-pole doppler filter being used to simulate the doppler shift present in the channel. Lastly, if either fading scenario (2) or (3) is chosen (Rician fading), the user is prompted to enter the specular-to-multipath power ratio (PWRRAT, in db). This is used to calculate the power or standard deviation of the multipath components, which are random Gaussian, statistically independent terms. Depending on this ratio, the values of the specular component and multipath components are calculated. It is important to note however, if the fading channel hasn't a specular component, then this question is not asked, since the fading is pure Rayleigh and every component has the same average power.

The general algorithm used in the simulation of the fading

channel (simulation of expression 6.1.6) is as follows:

```
BEGIN  /Fading Channel Routine/:
         < Initialize sum and index variables >
      DO < Faded sample generation  routine using  circular queue
           method >
         < Update the tap delay line >
         < Update the pointer to the oldest sample in the queue >
         < Set values for sample loop counters >
         < Obtain new tapped delay line multiplier coefficients >
        DO < Multiplication and accumulaion routine for the multi-
             path taps >
           < Perform operation of taps to the right of pointer >
           < Perform operation of taps to the left of pointer >
        IF < Interference is  faded then add the faded components
             of the interference >
        END < Multiplication and  accumulation  routine  for  the
              multipath taps >
      END < Faded sample generation routine using  circular queue
            method >
      END < Fading channel routine >.
```

6.2 UHF Uplink Satellite Channel

In the UHF uplink satellite channel subroutine, the transmitted signal is passed through a fading channel, if it exists in the uplink channel at that time (as prescribed by the user). If interference is present in the uplink, then it is possible for it to be either faded or not faded, even if the signal itself is faded. If it is faded, it is faded independently of the signal, and is then added onto the faded signal. The simulation prompts the user as to wether or not he wants the interference faded.

The algorithm for the UHF uplink satellite channel is as follows:

```
BEGIN / UHF uplink satellite channel routine /
 IF < Uplink fading channel is on >
 THEN
   DO < Fading channel routine >
      < Call fading channel >
   END< Fading channel routine >
 IF < Interference  is  present and faded, then add to faded
      signal >
 ELSE < IF  interference  is  present  and not faded, add to
        faded signal >
END  / UHF uplink satellite channel routine /
```

6.3 UHF Downlink Satellite Channel

The UHF downlink satellite channel performs primarily the same function as does the UHF uplink satellite channel, with the only exception being that the downlink has facilty for a space diversity receiver. If a space diversity receiver is present, than this routine allows for two seperate signals (i.e. each signal being faded independently of the other) coming into the receiver.

6.4 Gaussian noise generator

In many parts of the simulator, gaussian noise is needed. Of course, it is impossible to generate truly gaussian random variables using the computer but a good approximation to normal random variables is attanaible.

The method used consists in obtaining a pseudo random integer number distributed uniformly between 1 and 100 using the congruent generator method. The number obtained is used as a pointer to pick a number from a look-up table which contains numbers normally distributed The mean of these later numbers is zero and the variance is one. The table contains numbers between -3 and 3 ; therefore this is a limitation because a truly gaussian random variable can take any value between +oo and -oo . However this limitation is not really serious because a gaussian variable with a variance of 1 has a very small probability of taking values outside of the range +3 and -3. In general, this look-up table makes possible to generate gaussian random variables whose density function is very nearly gaussian near its mean, but the approximation, as it was stated before, is poorer for values in the tail of the distribution.

The algorithm is as follows:

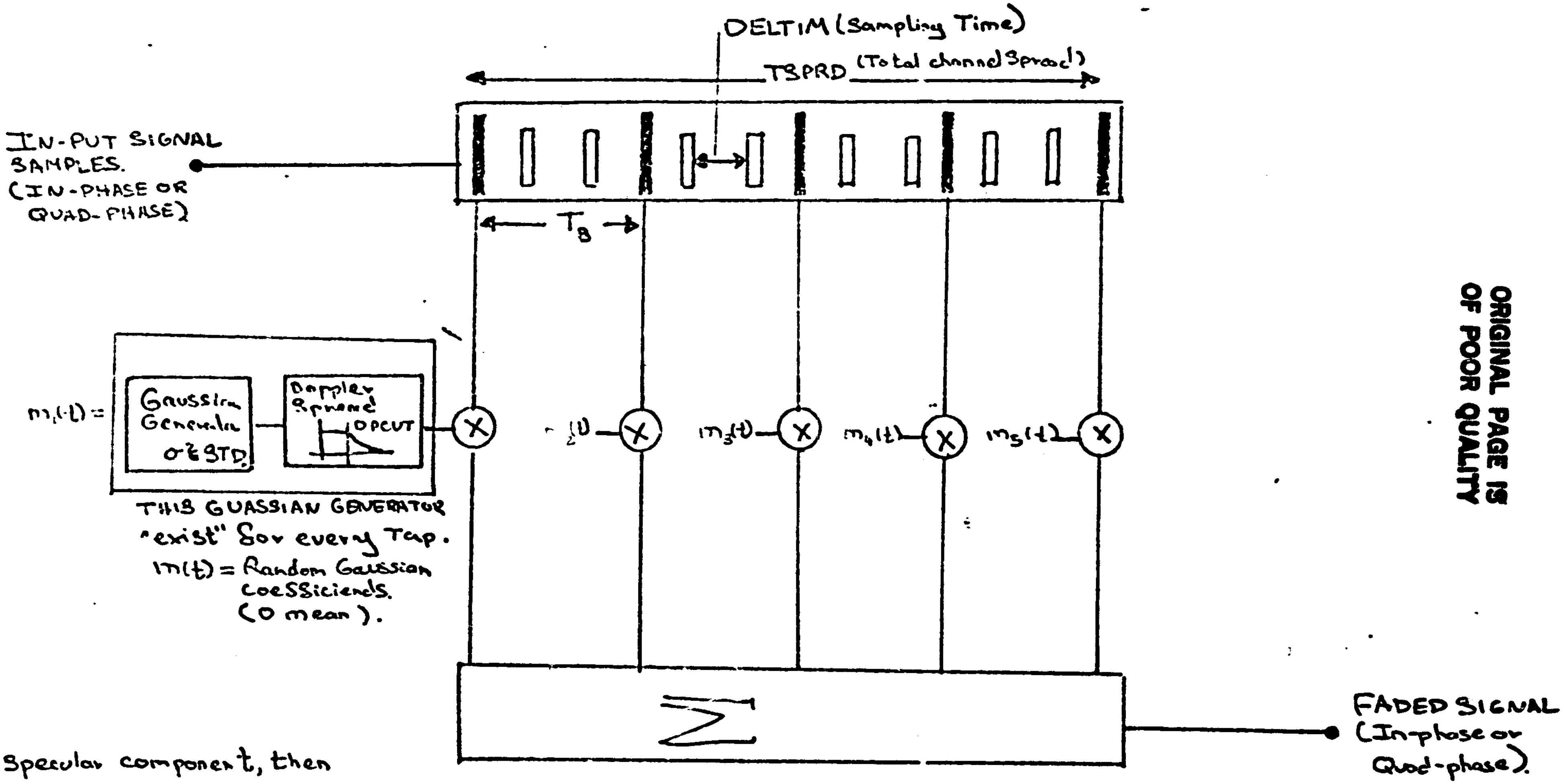
```
/Gaussian random generator/:
 < obtain a "random" number (integer) between 1 and 100 >
 < get a gaussian number from look-up table, the integer numbe
            is used as a pointer to this table >
END/Gaussian random generator/
```

6.5 The Satellite

The present version of the simulator does not simulate the communications satellite. The satellite is simulated by a stage of gain 1.

Fig. 6.1.1

Tap-Delay Line Model for Fading Channel



IS specular component, then

$m_1(t) = \sqrt{1 - \text{Power Ratio}}$

PWRRAT = Specular to Multipath power ratio in DB.

Power Ratio = PWRRAT converted from DB.

Multipath Coefficients have a Standard Deviation of: $m_i(t)$

$$STD = \sqrt{\frac{\text{Power Ratio}}{\#\text{ of Taps}}}$$

$T_B = 1/FMB$

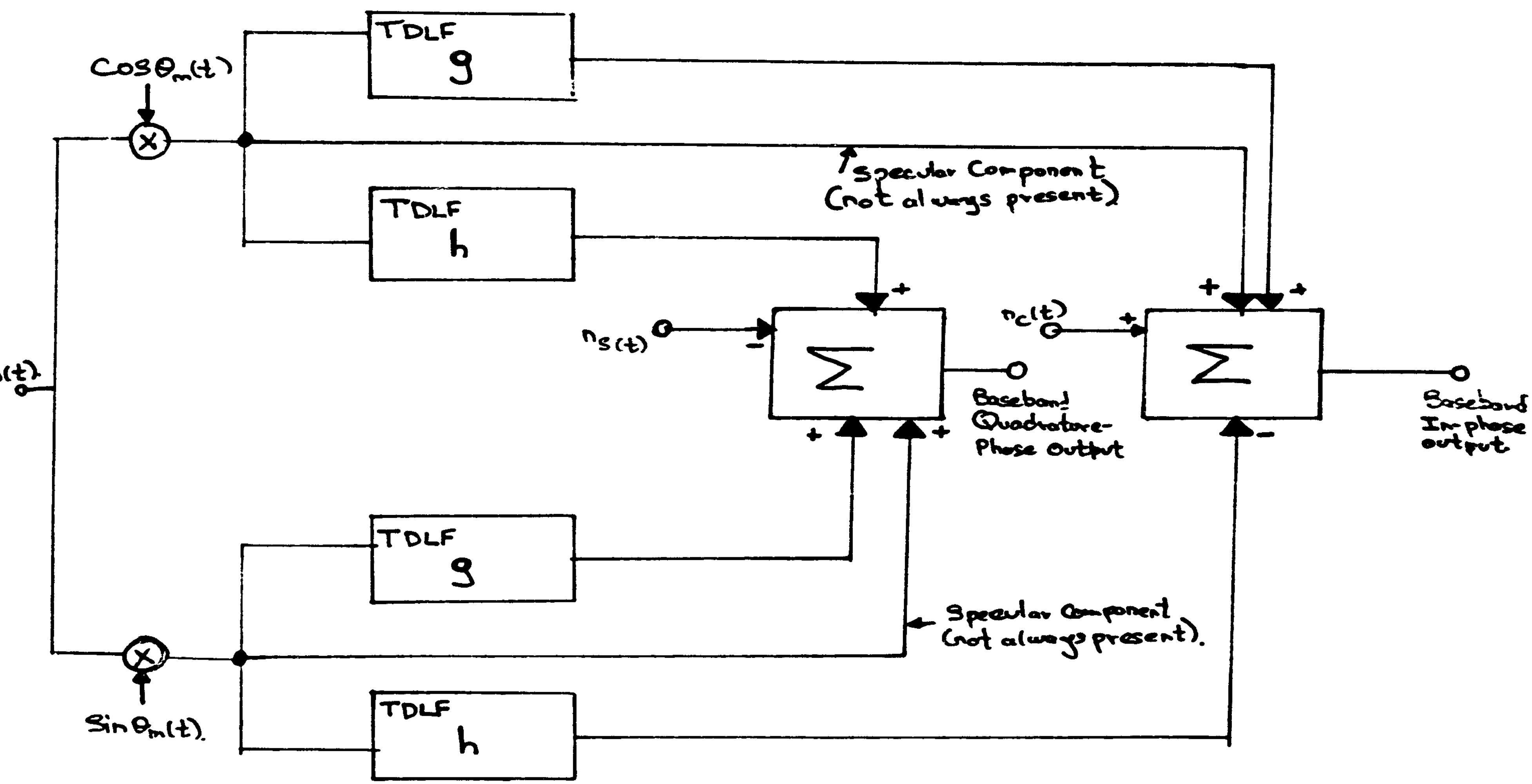
TSPRD = Multipath Spread time.

DPCUT = Doppler filter cutoff freq.

$\text{Number of Taps} = \frac{TSPRD}{T_B} + 1 = NT$

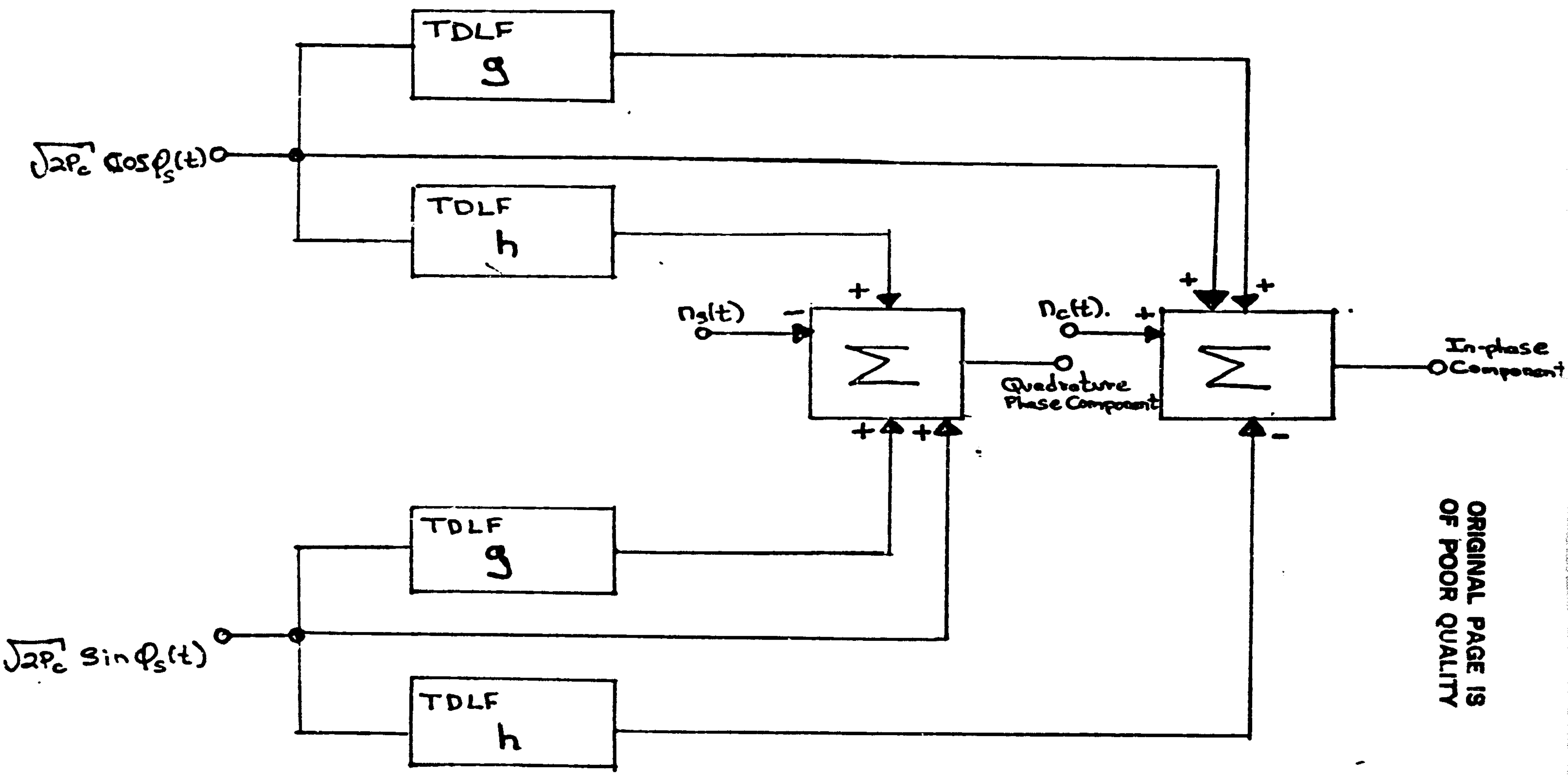
FIGURE 6.1.2

Fading Channel Simulater



Fading Channel Simulator (FM)

Figure 6.1.3



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TDLF - Tap - Delay - Line Filter

7. Interference Sources

The interference sources routine generates interference for both the uplink and downlink channels.

Parameters of the phase and carrier-to-interference power ratio are inputed. The carrier-to-interference power ratio is changed from db into a power by

POWINT=PC/(10**(CIRTIO/10.)) (7.1)

where

PC = power of the carrier,

CIRTIO = carrier-to-interference power ratio in db, and

POWINT = power of the interference,

The in-phase and quadrature-phase components of the interference are then calculated by the following algorithm:

RIS6P = POWINT * COS(PHASE) (7.2)

and

RIS6Q = POWINT * SIN(PHASE), (7.3)

where

PHASE = phase of the interference,

RIS6P = in-phase component of interference, and

RIS6Q = quadrature-phase component of interference.

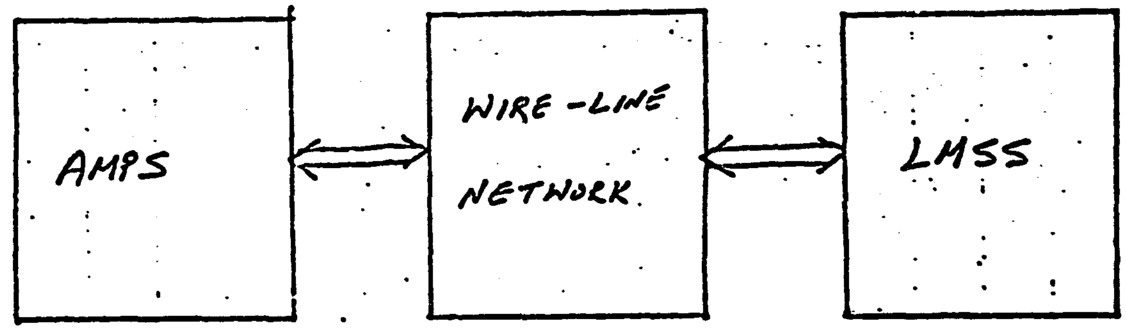

Figure 2.1.1 : Total Phone System

BASELINE COVERAGE
(18beams, 50-FOOT ANTENNA)

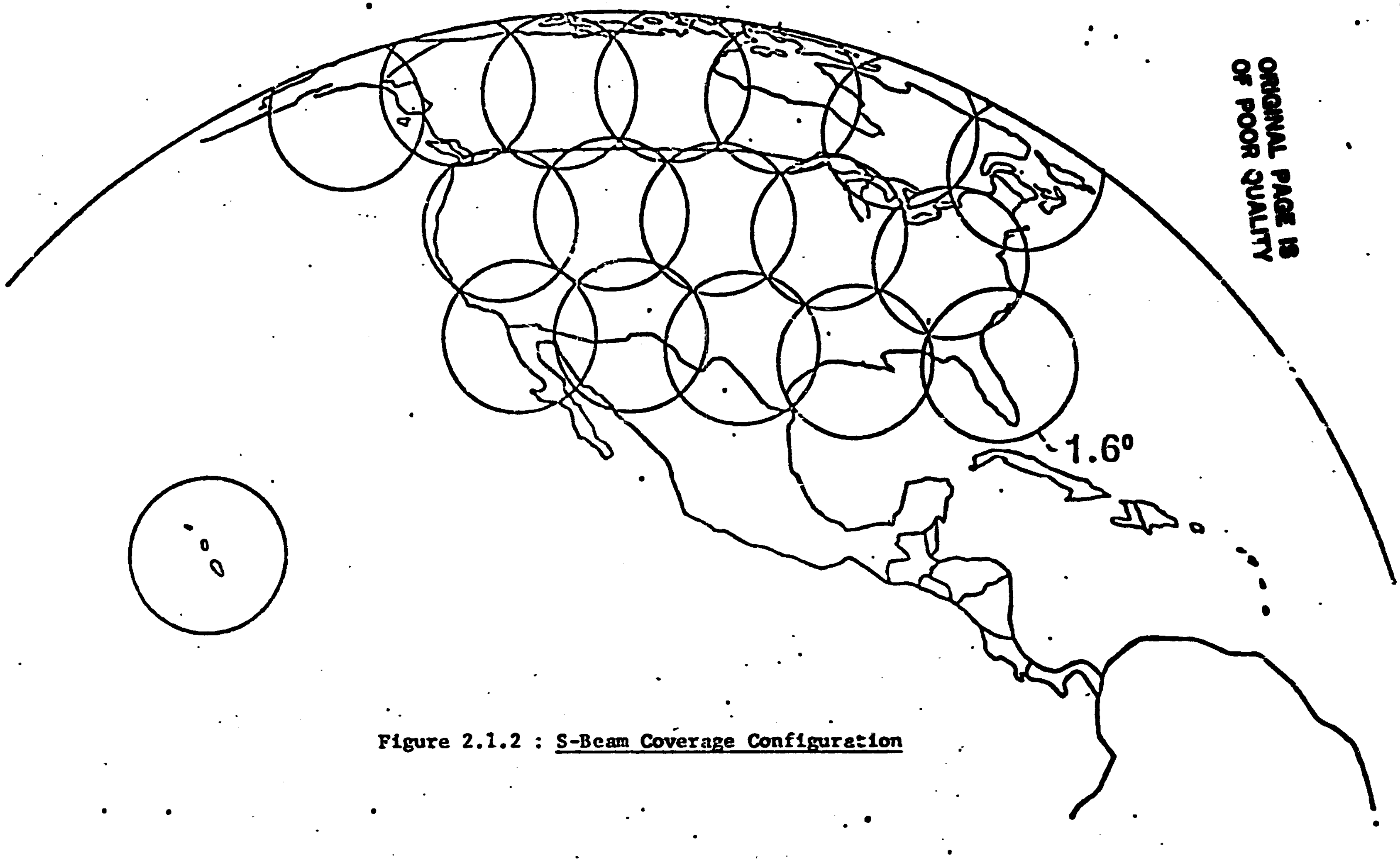

Figure 2.1.2 : S-Beam Coverage Configuration

HIGH—UHF SYSTEM CONFIGURATIONS

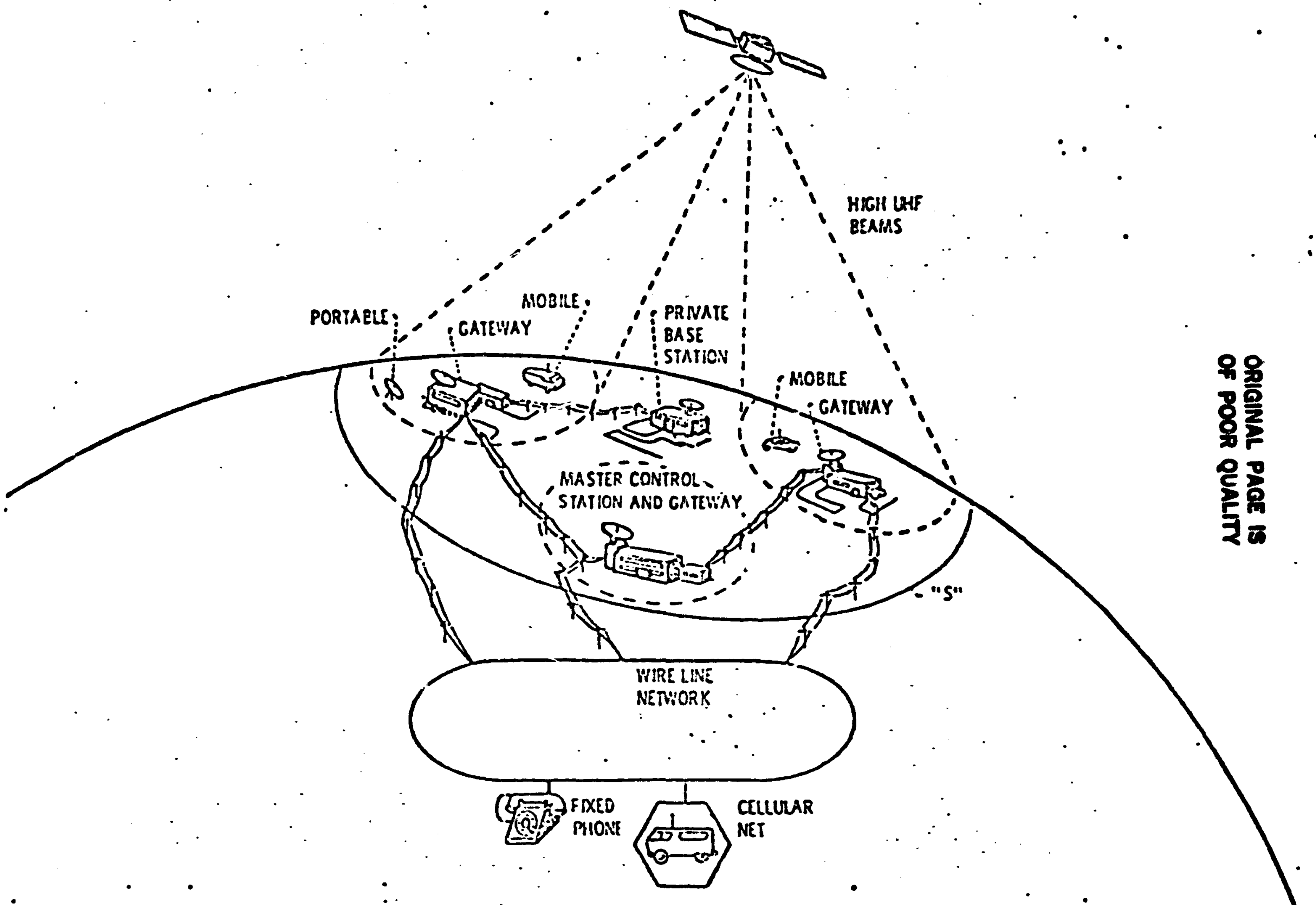

Figure 2.1.3 : LMSS System Configuration

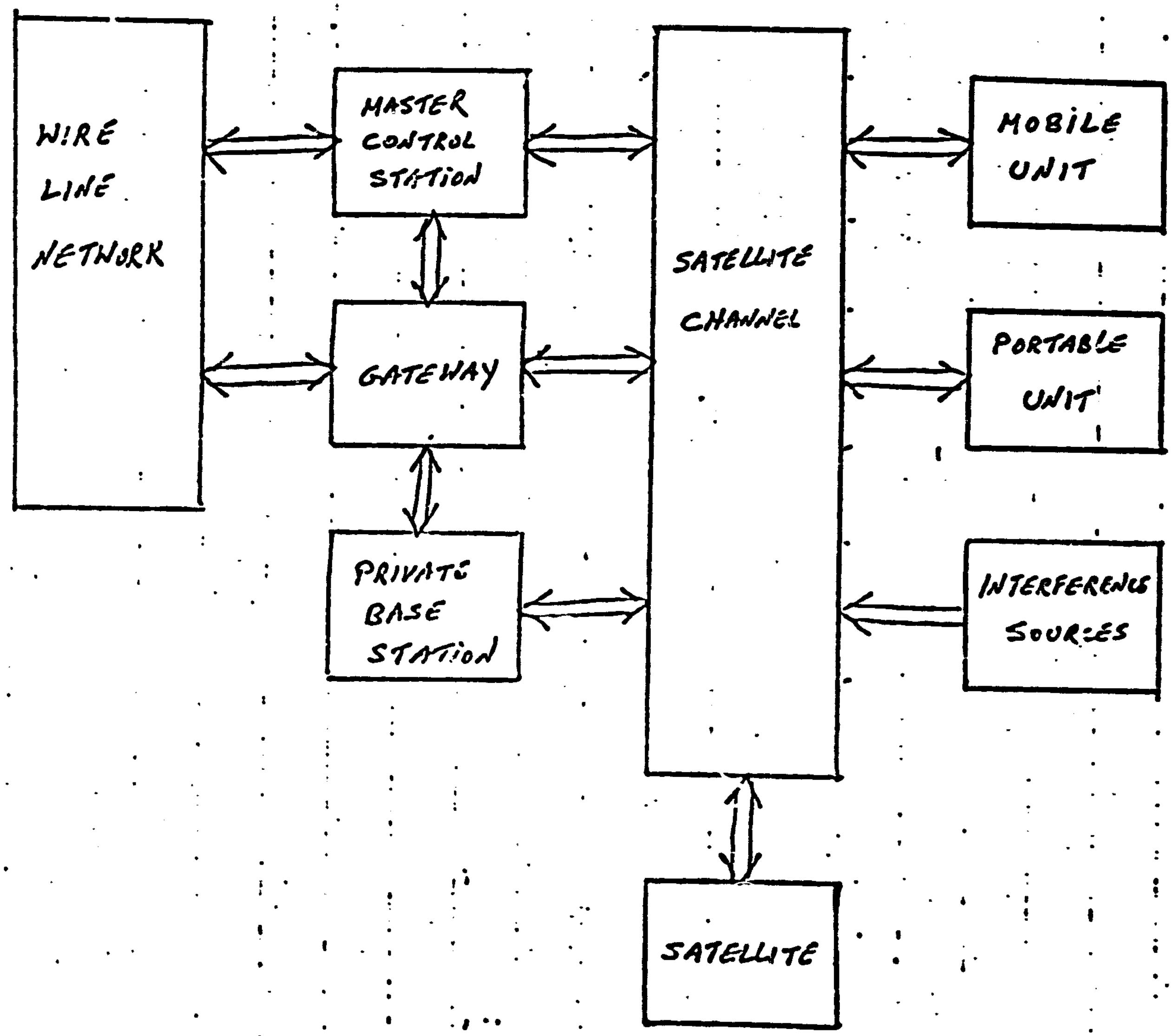

Figure 2.1.4 : LMSS Simulation Configuration

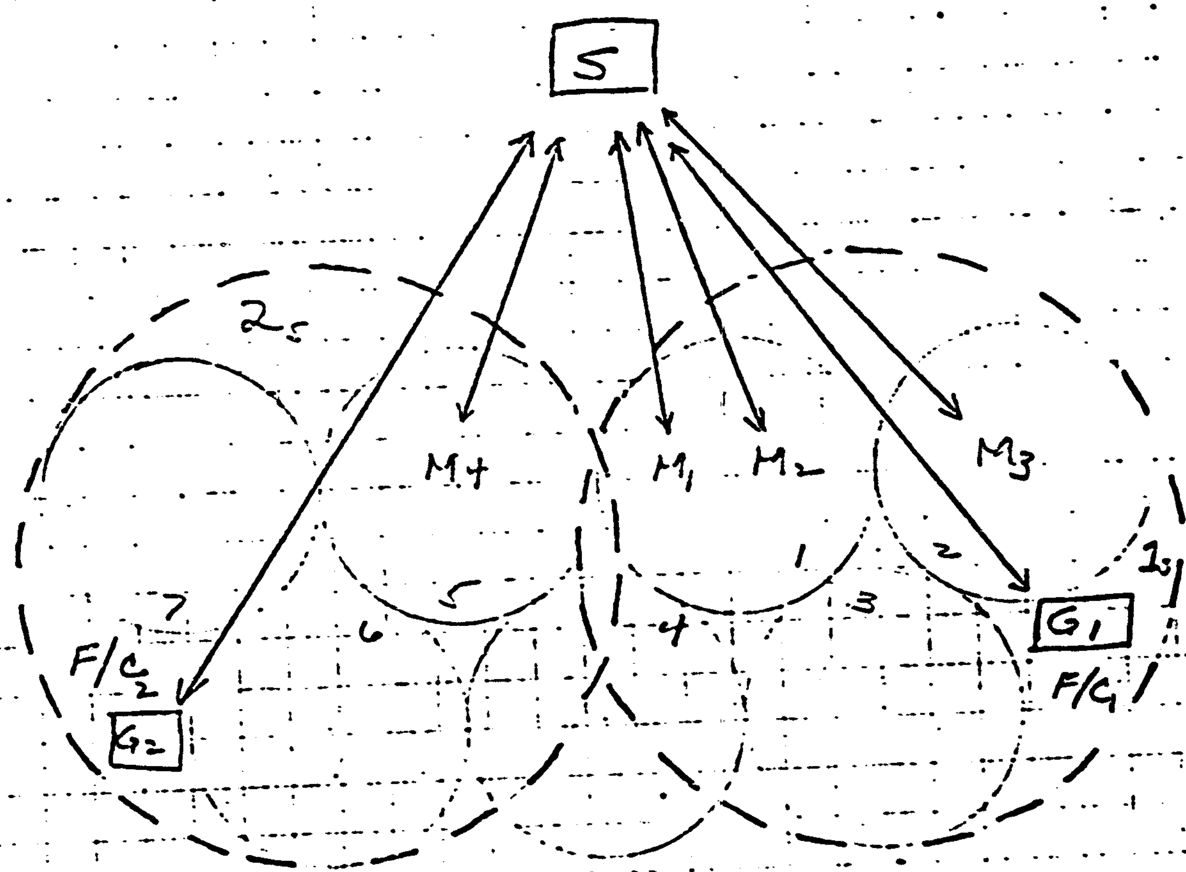

Figure 2.2.2 : Types of Call

---	S-Beam
___	UHF-Beam
G	Gateway
M	Mobile
F/C	Fixed / Cellular Phone
S	Satellite

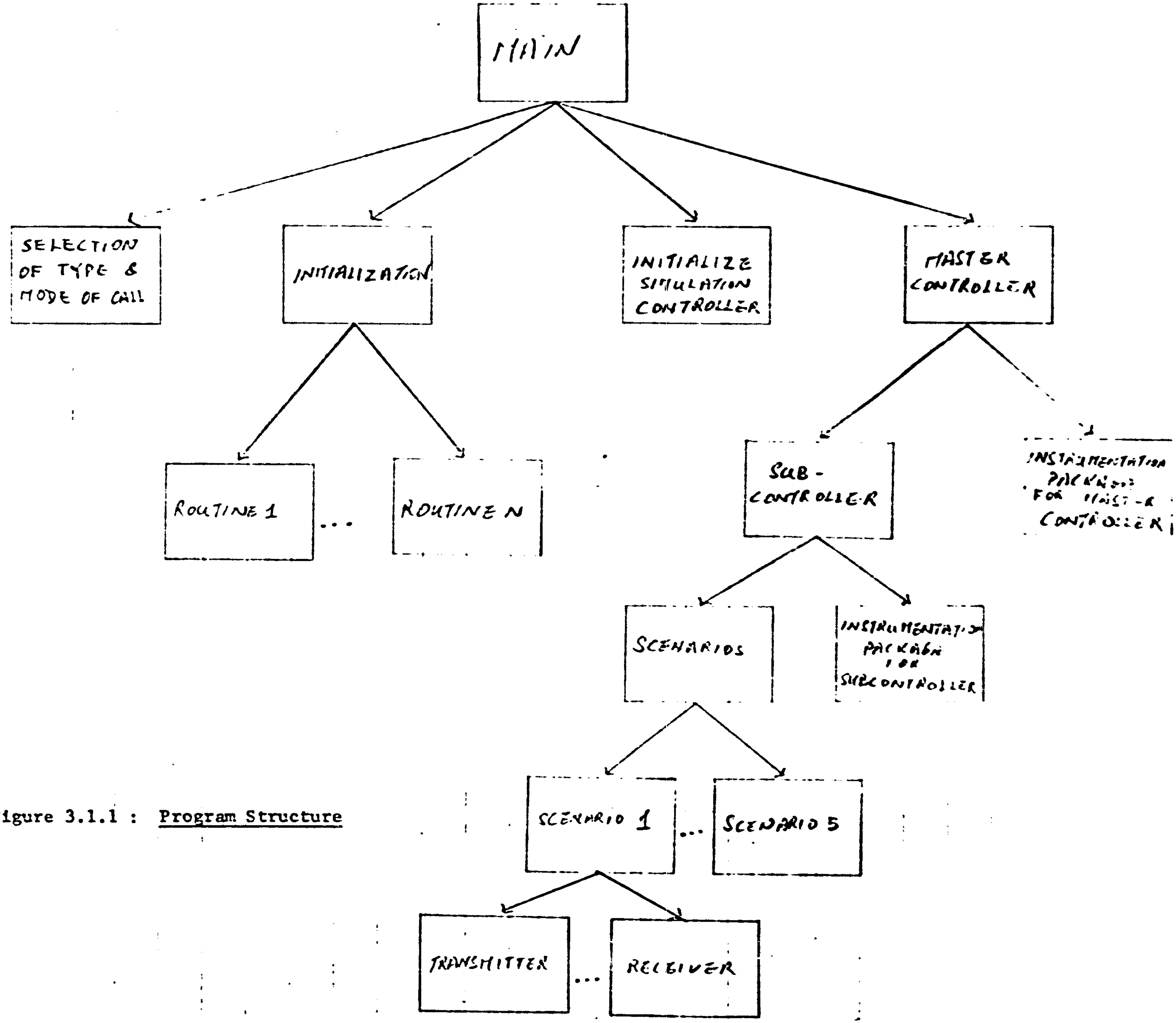

Figure 3.1.1 : Program Structure

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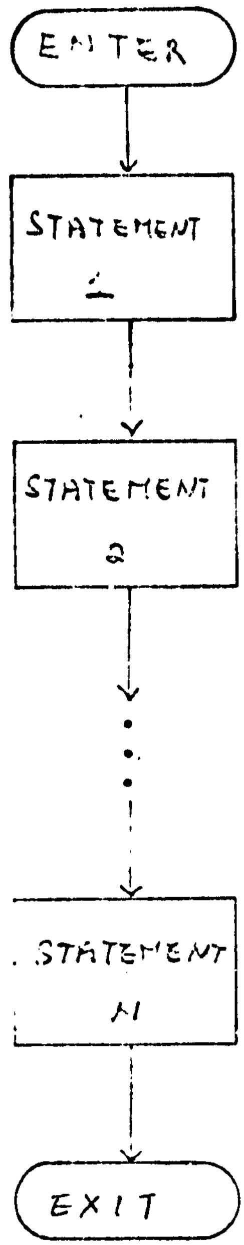

Figure 3.1.2 (a) : Block of Statements

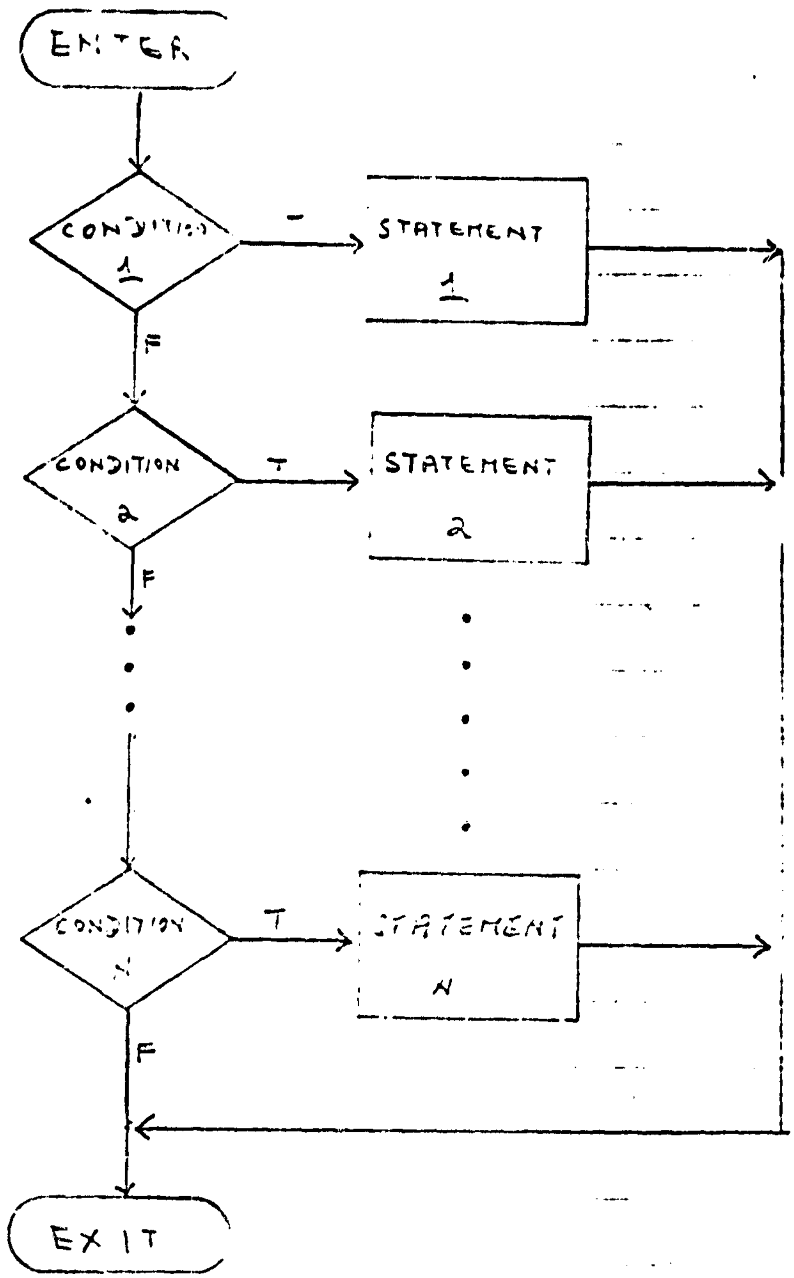

Figure 3.1.2 (b) : Selection Statement

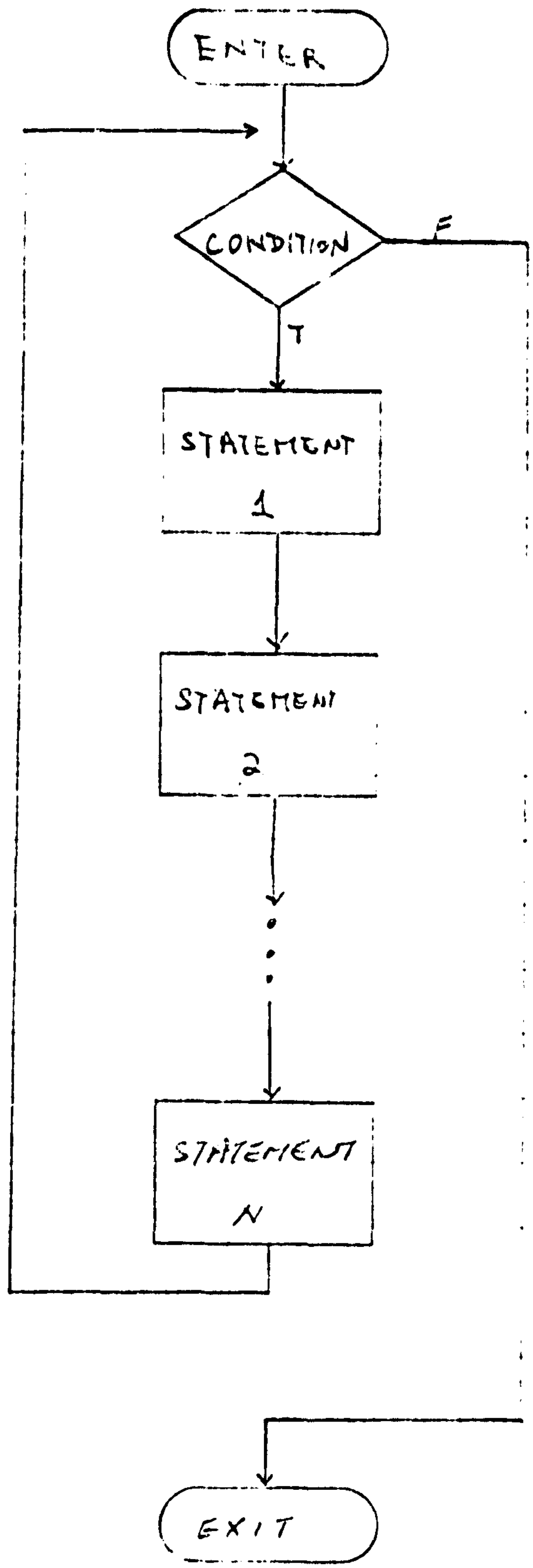

Figure 3.1.2 (c) : Pre-Test Loop

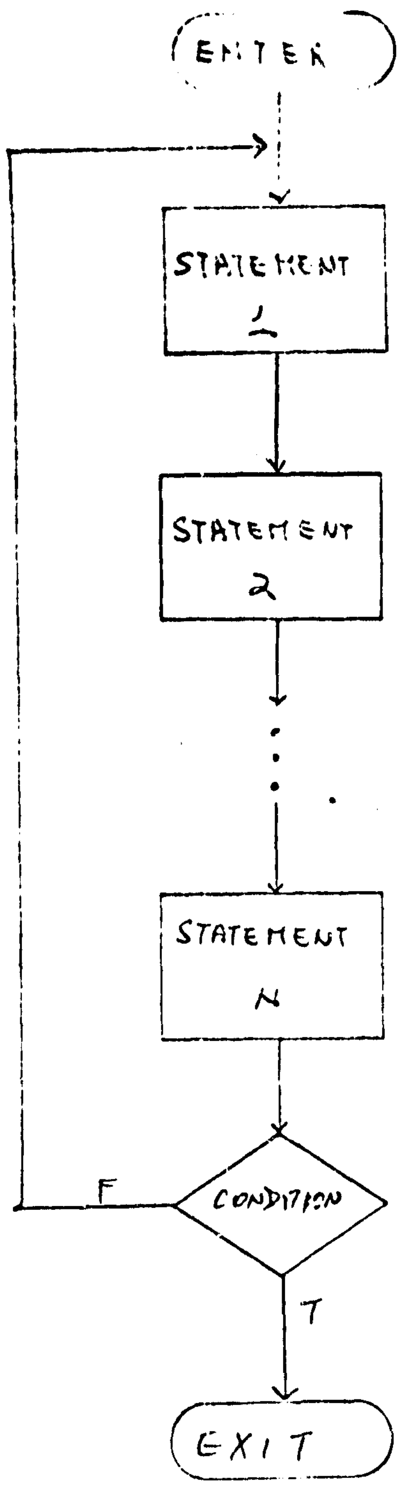

Figure 3.1.2 (d) : Post-Test Loop

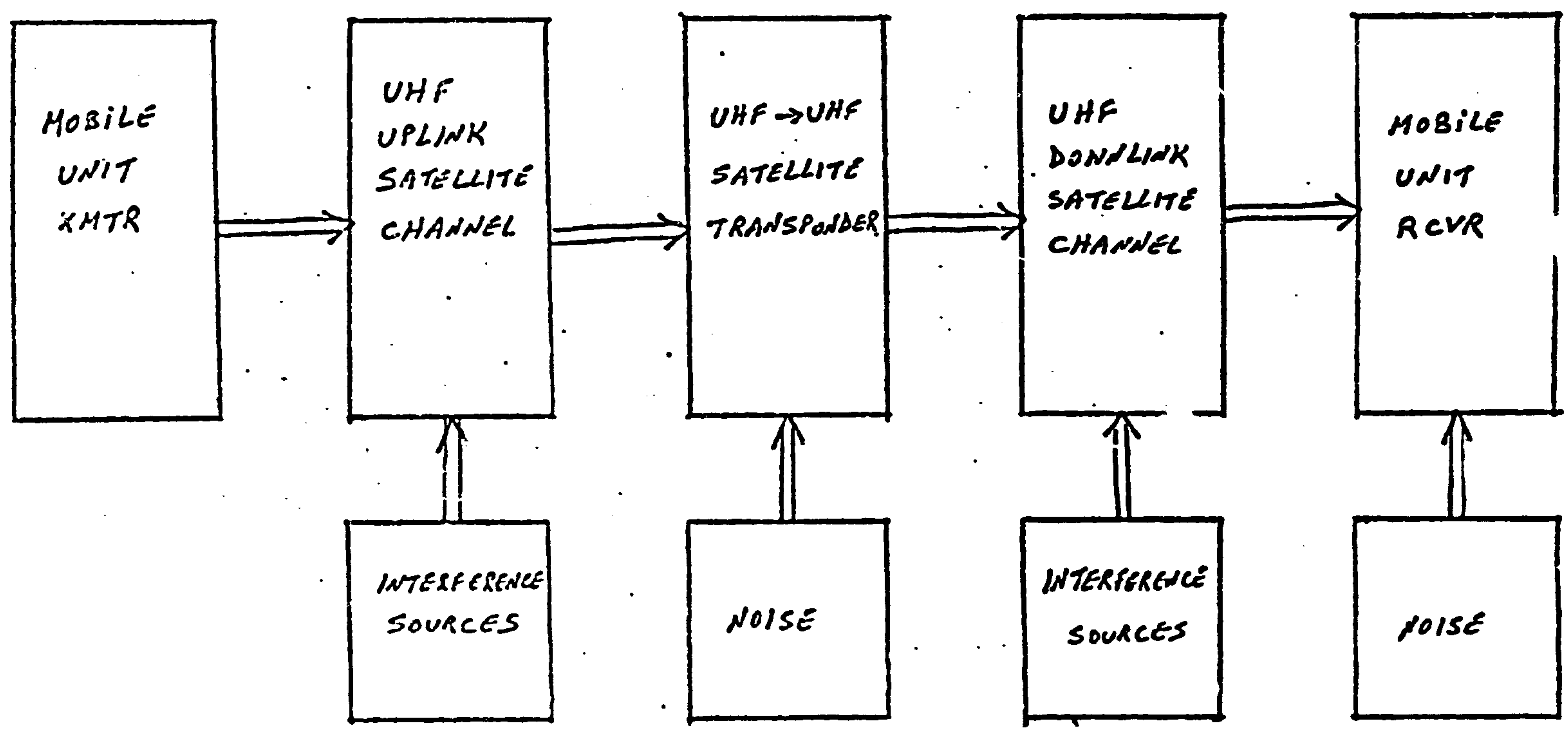

Figure 3.3.1 : Single Hop Single Gateway System

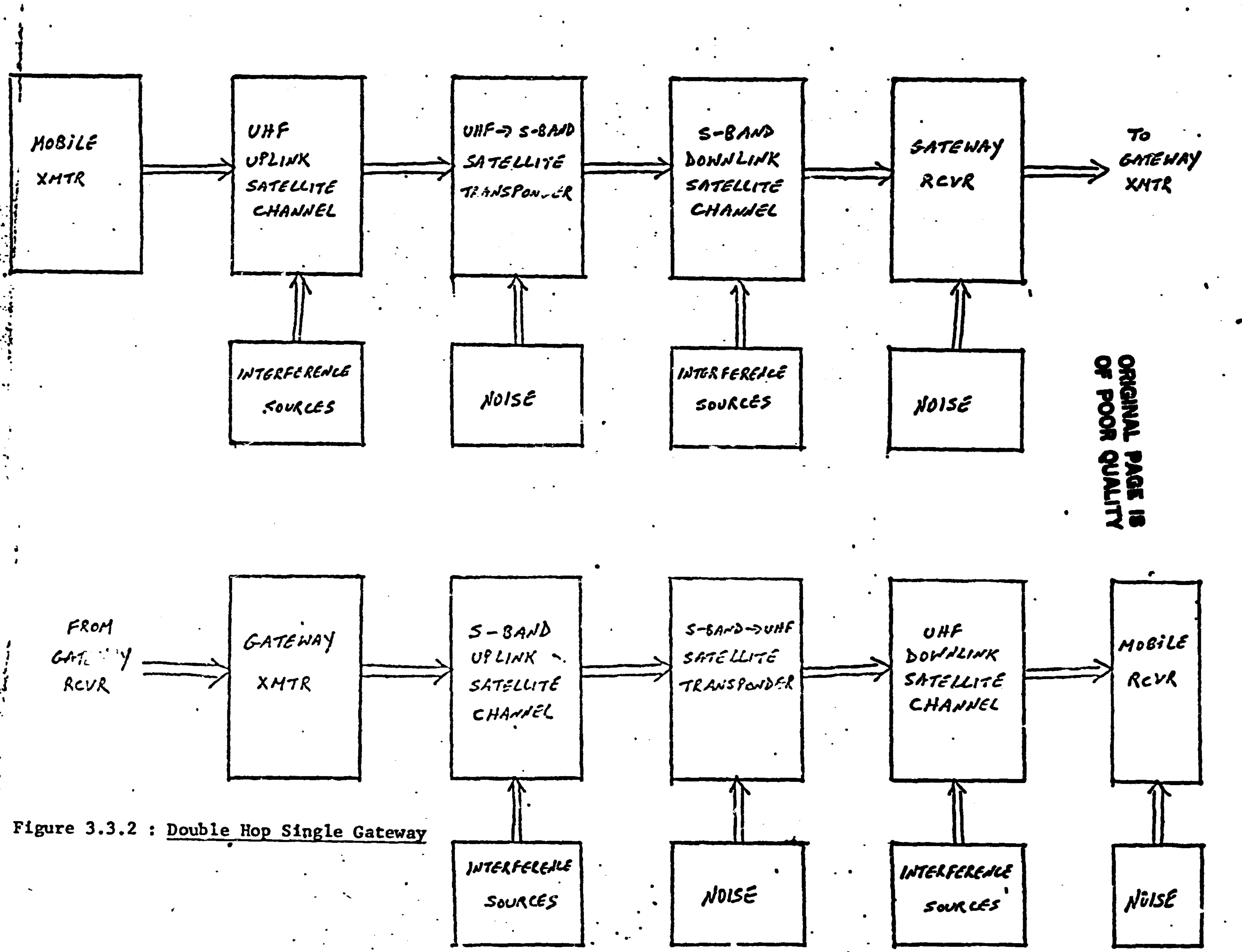

Figure 3.3.2 : Double Hop Single Gateway

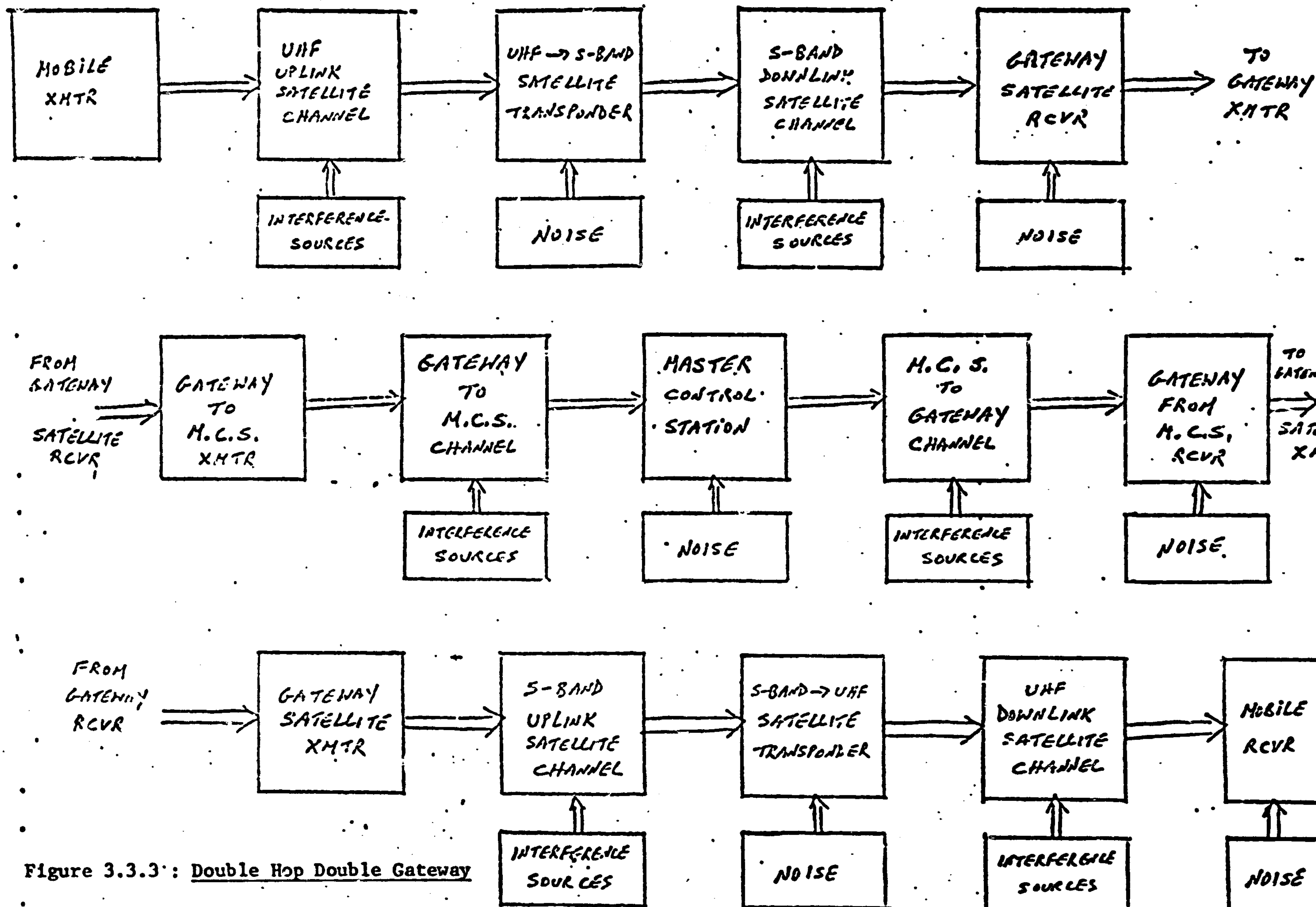

Figure 3.3.3 : Double Hop Double Gateway

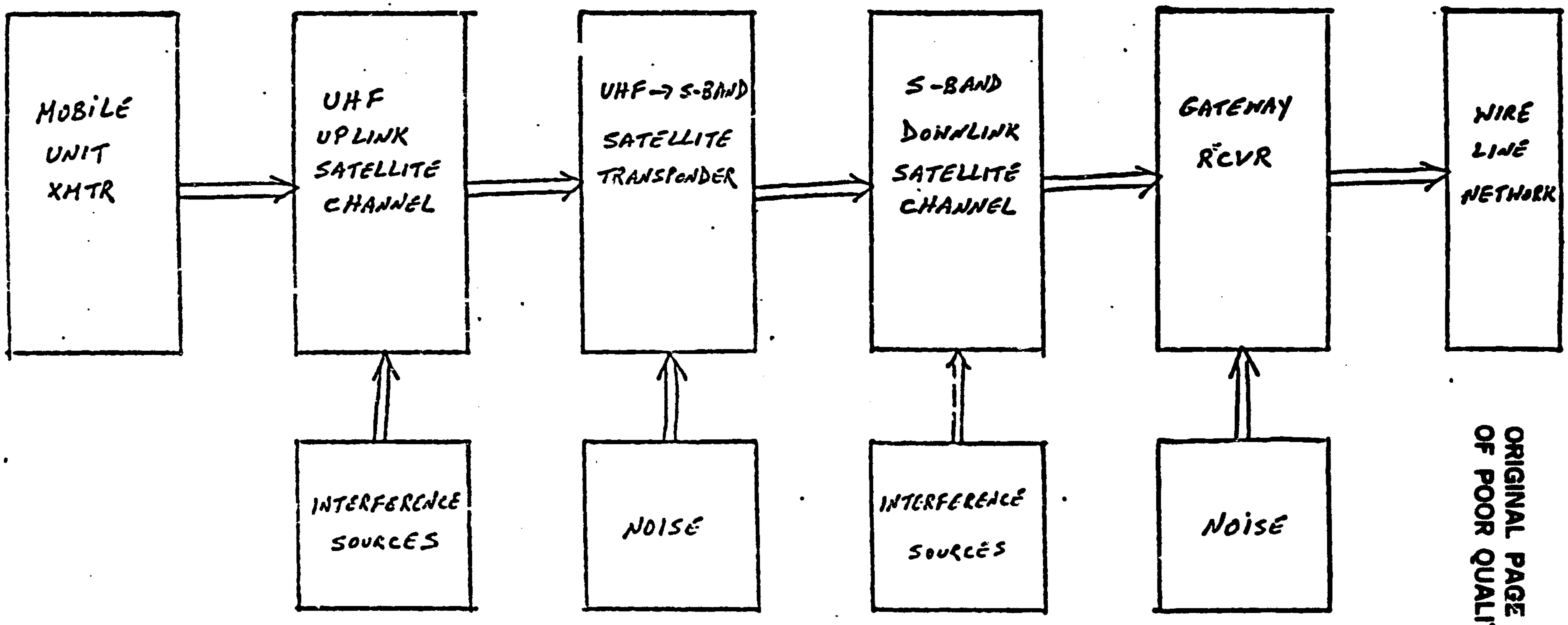

Figure 3.3.4 : Mobile to Wireline System

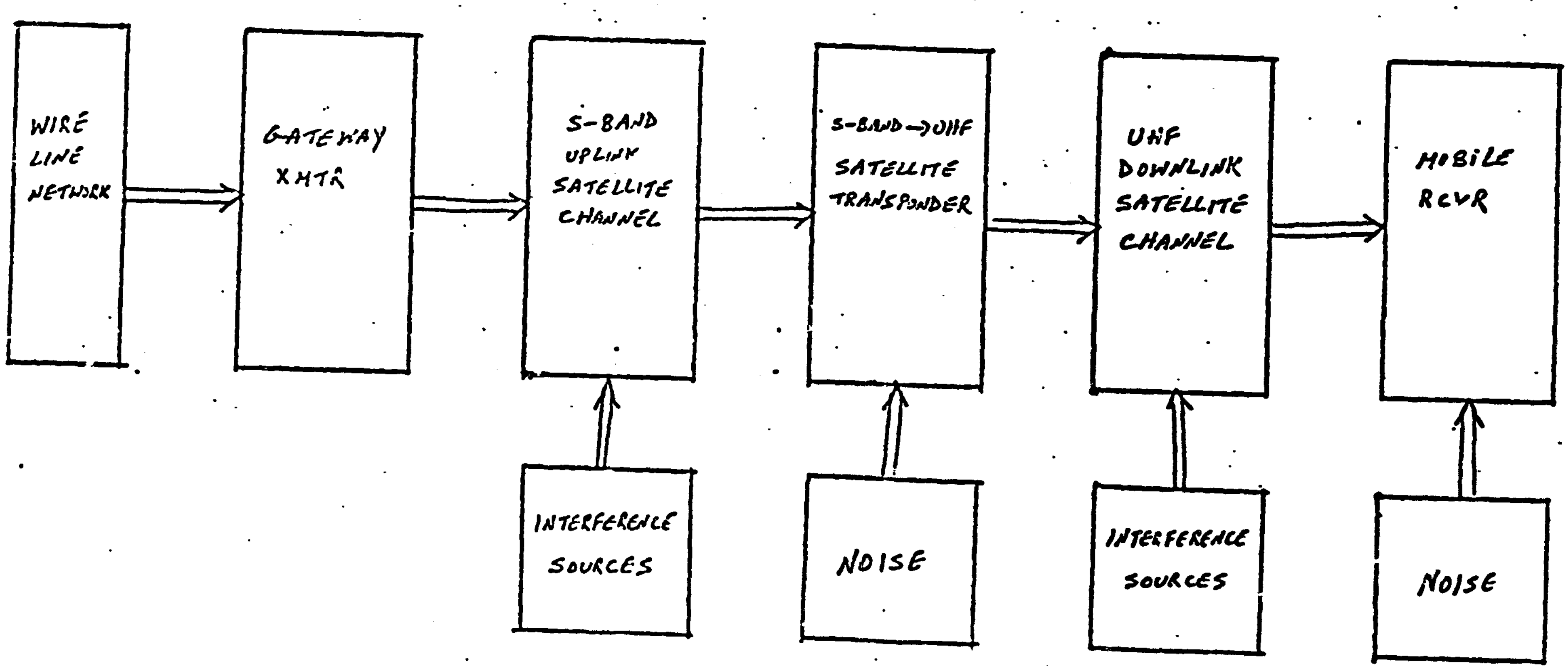

Figure 3.3.5 : Wireline to Mobile System

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FIG. 6.1.4

```
TYPE OF CALL AVAILABLE:
1: M1->M2, RURAL MOBILE TO RURAL MOBILE IN SAME UHF  EAM
2: M1->M3, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN SAME S-BAND
3: M1->M4, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN DIFFERENT S-BAND
4: M1->FC1, RURAL MOBIL_ TO FIXED IN SAME S-BAND BEA
5: M1->FC2, RURAL MOBILE TO FIXED IN DIFFERENT S-BAN  BEAM
6: FC1->M1, FIXED TO RURAL MOBILE IN SAME S-BAND BEA
7: FC1->M1, FIXED TO RURAL MOBILE IN DIFFERENT S-BAN  BEAM


INPUT TYPE OF CALL TO BE SIMULATED: 1


MODE OF CALL AVAILABLE:
1: M1->M2, HARD WIRED TRANSPONDER
2: M1->M2, DIRECT SWITCHED TRANSPONDER
3: M1->M2, INDIRECT SWITCHED TRANSPONDER
4: M1->S1->M2, DOUBLE HOP SYSTEM


INPUT MODE OF CALL TO BE SIMULATED: 1
FREQUENCY OF THE BASEBAND SIGNAL ( LESS THAN 3000 HZ  ) (IN HERTZ  F7.2): 1000.00 HZ
NOTE : POWER OF THE BASEBAND SIGNAL HAS TO BE LESS O  EQUAL THAN .5 WATTS;
       SO THAT THE MAXIMUM INSTANTANEOUS FREQUENCY D  VIATION IS
       LESS OR EQUAL THAN 12 00 HZ
POWER OF THE BASEBAND SIGNAL(IN WATTS, F7.2):    0.50 WATTS
THE CARRIER POWER (IN WATTS, F7.2):   1.00 WATTS
THE FREQUENCY DEVIATION (IN HERTZ, F6.2): 12.00 HZ.  (FIXED FOR NOW)


HOW MANY TIMES THE NYQUIST RATE DO YOU WANT THE SAMP ING FREQUENCY TO BE? (2 - 4, 11): 2
THE CARRIER FREQUENCY (IN HERTZ, F7.2):(NOT USED FOR NOW)

DO YOU WANT THE COMPRESSOR/EXPANDER IN?  Y-YES , N-N  : Y

DO YOU WANT THE PRE-EMPHASIS - DE-EMPHASIS FILTERS I  ?  Y-YES , N-NO : Y

DO YOU WANT A CO-CHANNEL INTERFERER IN THE UPLINK? Y YES,N-NO: N

IS FADING PRESENT IN THE UPLINK   CHANNEL? Y-YES,N-N : N

DO YOU WANT A CO-CHANNEL INTERFERER IN THE DOWNLINK? Y-YES,N-NO: N

IS FADING PRESENT IN THE DOWNLINK CHANNEL? Y-YES,N-N : Y

IS A SP    DIVERSITY RECEIVER BEING USED? Y-YES,N-NO  N
```

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```
SET PARAMETERS FOR FADING CHANNEL.

TYPES OF FADING CHANNELS AVAILABLE:
 1: NO SPECULAR COMPONENT (RALEIGH FADING)
 2: SPECULAR COMPONENT, SHORTEST PATH
 3: SPECULAR COMPONENT, MEAN PATH
 INPUT TYPE OF CHANNEL: 2

ENTER THE MULTIPATH SPREAD TIME (IN MICROSECONDS, F9 2):  0.500000E 03

ENTER THE DOPPLER SPREAD BANDWIDTH (IN HERTZ, F7.2):   1.00 HZ

ENTER THE SPECULAR-TO-MULTIPATH POWER RATIO (IN DB, I3): 100 DB

ARE MULTIPLE SNR VALUES TO BE TESTED? Y-YES,N-NO: Y
```

Fig. 6.1.4

```
INPUT THE RANGE OF SNR VALUES TO BE TESTED (IN DB):
INPUT THE INITIAL VALUE OF SNR (-99 -- +99, I3):-1. B
INPUT THE INCREMENT VALUE ( 1 -- 99, I2): 2 DB
INPUT THE ENDING VALUE OF SNR (-99 -- +99, I3):  40 B
IS SPACE DIVERSITY RECEIVER REQUIRED? (Y/N)  N

INPUT THE APPROXIMATE DURATION OF SIMULATION IN SECONDS (0.01 - 9.99, F4.2): 0.10 SEC.

TYPE OF PERFORMANCE MEASUREMENT AVAILABLE:
  1. COMPARE RECOVERED OUTPUT SIGNAL TO ORIGINAL INPUT SIGNAL
  2. MEASURE OUTPUT SIGNAL TO NOISE RATIO

INPUT TYPE:2
```

FIG. 6.6.1.4

INPUT SNR (DB)	OUTPUT SNR (D)
-10.0000000	0.4917583
-8.0000000	0.8644104
-6.0000000	1.4285517
-4.0000000	2.5974664
-2.0000000	4.5603704
0.0	6.1481265
2.0000000	9.4655837
4.0000000	15.0329665
6.0000000	34.2338562
8.0000000	39.5905417
10.0000000	41.8656616
12.0000000	44.0155945
14.0000000	45.9294566
16.0000000	48.6082458
18.0000000	49.7747040
20.0000000	52.6611328
22.0000000	54.0501662
24.0000000	55.6845245
26.0000000	58.0103760
28.0000000	60.1772003
30.0000000	61.9803619
32.0000000	64.1505432
34.0000000	65.5447388
36.0000000	63.0054626
38.0000000	69.9577923
40.0000000	71.6414795

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I SNR (DB)	O SNR (DB)
-10.000000	-[illegible].6395155
-8.000000	-[illegible].627[illegible] 49
-6.000000	[illegible].3395[illegible]63
-4.000000	1.172[illegible]77
-2.000000	2.[illegible]6[illegible]832[illegible]
.000	2.929[illegible]724
2.000000	4.973[illegible]998
4.000000	8.3063826
6.000000	13.1725788
8.000000	22.438 798
10.000000	37.29[illegible]8783
12.000000	4[illegible].934[illegible]985
14.000000	42.156 374
16.000000	44.91[illegible]3748
18.000000	46.7586 6[illegible]
20.000000	48.7257[illegible]74
22.000000	5[illegible].922[illegible]58[illegible]
24.000000	52.9429932
26.000000	54.7192[illegible]88
28.000000	57.3687266
30.000000	58.8286743
32.000000	6[illegible].82[illegible]5[illegible]14
34.000000	62.71 5[illegible]65
36.000000	64.522[illegible]593
38.000000	66.76[illegible]8645
40.000000	68.9667[illegible]69

FIG. 6.1.6

```
TYPE OF CALL AVAILABLE:
1: M1->M2, RURAL MOBILE TO RURAL MOBILE IN SAME UHF  EAM
2: M1->M3, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN SAME S-BAND
3: M1->M4, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN DIFFERENT S-BAND
4: M1->FC1, RURAL MOBILE TO FIXED IN SAME S-BAND BEA
5: M1->FC2, RURAL MOBILE TO FIXED IN DIFFERENT S-BAN  BEAM
6: FC1->M1, FIXED TO RURAL MOBILE IN SAME S-BAND BEA
7: FC2->M1, FIXED TO RURAL MOBILE IN DIFFERENT S-BAN  BEAM


INPUT TYPE OF CALL TO BE SIMULATED: 1


MODE OF CALL AVAILABLE:
1: M1->M2, HARD WIRED TRANSPONDER
2: M1->M2, DIRECT SWITCHED TRANSPONDER
3: M1->M2, INDIRECT SWITCHED TRANSPONDER
4: M1->S1->M2, DOUBLE HOP SYSTEM


INPUT MODE OF CALL TO BE SIMULATED: 1
FREQUENCY OF THE BASEBAND SIGNAL ( LESS THAN 3[illegible] HZ  ) (IN HERTZ, F7.2): 1000.00 HZ
NOTE : POWER OF THE BASEBAND SIGNAL HAS TO BE LESS O  EQUAL THAN .5 WATTS;
       SO THAT THE MAXIMUM INSTANTANEOUS FREQUENCY D VIATION IS
       LESS OR EQUAL THAN 12.00 HZ
POWER OF THE BASEBAND SIGNAL(IN WATTS, F7.2):    .50 WATTS
THE CARRIER POWER (IN WATTS, F7.2):   1.00 WATTS
THE FREQUENCY DEVIATION (IN HERTZ, F6.2): 12.00 HZ.  (FIXED FOR NOW)


HOW MANY TIMES THE NYQUIST RATE DO YOU WANT THE SAMP ING FREQUENCY TO BE? (2 - 4, 11): 2
THE CARRIER FREQUENCY (IN HERTZ, F7.2):(NOT USED FOR NOW)

DO YOU WANT THE COMPRESSOR/EXPANDER IN?  Y-YES , N-N  : Y

DO YOU WANT THE PRE-EMPHASIS - DE-EMPHASIS FILTERS I  ?  Y-YES , N-NO : Y

DO YOU WANT A CO-CHANNEL INTERFERER IN THE UPLINK? Y YES,N-NO: N

IS FADING PRESENT IN THE UPLINK   CHANNEL? Y-YES,N-N : N

DO YOU WANT A CO-CHANNEL INTERFERER IN THE DOWNLINK? Y-YES,N-NO: N

IS FADING PRESENT IN THE DOWNLINK CHANNEL? Y-YES,N-N : Y

IS A SPACE DIVERSITY RECEIVER BEING USED? Y-YES,N-NO  N
THE DOWNLINK FADING CHANNEL IS PRESENT
```

```
SET PARAMETERS FOR FADING CHANNEL.

TYPES OF FADING CHANNELS AVAILABLE:
 1: NO SPECULAR COMPONENT (RALEIGH FADING)
 2: SPECULAR COMPONENT, SHORTEST PATH
 3: SPECULAR COMPONENT, MEAN PATH
 INPUT TYPE OF CHANNEL: 2

ENTER THE MULTIPATH SPREAD TIME (IN MICROSECONDS, F9 2):  0.5000000E 03

ENTER THE DOPPLER SPREAD BANDWIDTH (IN HERTZ, F7.2):   1.00 HZ

ENTER THE SPECULAR-TO-MULTIPATH POWER RATIO (IN DB, I3):  35 DB

ARE MULTIPLE SNR VALUES TO BE TESTED? Y-YES,N-NO: Y
```

FIG.6.1.6.

```
INPUT THE RANGE OF SNR VALUES TO BE TESTED (IN DB):
INPUT THE INITIAL VALUE OF SNR (-99 -- +99, I3):-1. .B
INPUT THE INCREMEN. VALUE (01 -- 99, I2): 2 DB
INPUT THE ENDING VALUE OF SNR (-99 -- +99, I3):  40 B
IS SPACE DIVERSIFY RECEIVER REQUIRED? (Y/N)  N


INPUT THE APPROXIMATE DURATION OF SIMULATION IN SECONDS (0.01 - 9.99, F4.2): 0.10 SEC.


TYPE OF PERFORMANCE MEASUREMENT AVAILABLE:
  1. COMPARE RECOVERED OUTPUT SIGNAL TO ORIGINAL INPUT SIGNAL
  2. MEASURE OUTPUT SIGNAL TO NOISE RATIO

INPUT TYPE:2
```

FIG. 6.1.6.

INPUT SNR (DB)	OUTPUT SNR (D)
-10.0000000	0.4507501
-8.0000000	0.8521128
-6.0000000	1.3745985
-4.0000000	2.7256918
-2.0000000	4.6239424
0.0	6.0571175
2.0000000	9.4529257
4.0000000	14.6275043
6.0000000	34.2102814
8.0000000	39.5794983
10.0000000	41.8896484
12.0000000	44.0170288
14.0000000	45.8502350
16.0000000	48.5830383
18.0000000	49.7373962
20.0000000	52.6339417
22.0000000	53.9416962
24.0000000	55.5568542
26.0000000	57.7361145
28.0000000	59.8244781
30.0000000	61.4869690
32.0000000	63.8338470
34.0000000	65.7401733
36.0000000	66.0486755
38.0000000	68.3430786
40.0000000	70.3086395

FIG. 6.2.6


```
TYPE OF CALL AVAILABLE:
1: M1->M2, RURAL MOBILE TO RURAL MOBILE IN SAME UHF  EAM
2: M1->M3, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT JHF BEAM, IN SAME S-BAND
3: M1->M4, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN DIFFERENT S-BAND
4: M1->FC1, RURAL MOBILE TO FIXED IN SAME S-BAND BEA
5: M1->FC2, RURAL MOBILE TO FIXED IN DIFFERENT S-BAN  BEAM
6: FC1->M1, FIXED TO RURAL MOBILE IN SAME S-BAND BEA
7: FC2->M1, FIXED TO RURAL MOBILE IN DIFFERENT S-BAN  BEAM


INPUT TYPE OF CALL TO BE SIMULATED: 1


MODE OF CALL AVAILABLE:
1: M1->M2, HARD WIRED TRANSPONDER
2: M1->M2, DIRECT SWITCHED TRANSPONDER
3: M1->M2, INDIRECT SWITCHED TRANSPONDER
4: M1->G1->M2, DOUBLE HOP SYSTEM


INPUT MODE OF CALL TO BE SIMULATED: 1
FREQUENCY OF THE BASEBAND SIGNAL ( LESS THAN 3000 HZ  ) (IN HERTZ, F7.2): 1000.00 HZ
NOTE : POWER OF THE BASEBAND SIGNAL HAS TO BE LESS O  EQUAL THAN .5 WATTS:
       SO THAT THE MAXIMUM INSTANTANEOUS FREQUENCY D VIATION IS
       LESS OR EQUAL THAN 12000 HZ
POWER OF THE BASEBAND SIGNAL(IN WATTS, F7.2):    0.50 WATTS
THE CARRIER POWER (IN WATTS, F7.2):    1.00 WATTS
THE FREQUENCY DEVIATION (IN HERTZ, F8.2): 12000 HZ.  (FIXED FOR NOW)


HOW MANY TIMES THE NYQUIST RATE DO YOU WANT THE SAMP ING FREQUENCY TO BE? (2 - 4, 11): 2
THE CARRIER FREQUENCY (IN HERTZ, F7.2):(NOT USED FOR NOW)

DO YOU WANT THE COMPRESSOR/EXPANDER IN?  Y-YES , N-N  : Y

DO YOU WANT THE PRE-EMPHASIS - DE-EMPHASIS FILTERS I  ?  Y-YES , N-NO : Y

DO YOU WANT A CO-CHANNEL INTERFERER IN THE UPLINK? Y YES,N-NO: N

IS FADING PRESENT IN THE UPLINK   CHANNEL? Y-YES,N-N : N

DO YOU WANT A CO-CHANNEL INTERFERER IN THE DOWNLINK? Y-YES,N-NO: N

IS FADING PRESENT IN THE DOWNLINK CHANNEL? Y-YES,N-N : Y

IS A SPACE DIVERSITY RECEIVER BEING USED? Y-YES,N-NO  Y
THE DOWNLINK FADING CHANNEL IS PRESENT
```

```
SET PARAMETERS FOR FIRST FADING CHANNEL:

TYPES OF FADING CHANNELS AVAILABLE:
 1: NO SPECULAR COMPONENT (RALEIGH FADING)
 2: SPECULAR COMPONENT, SHORTEST PATH
 3: SPECULAR COMPONENT, MEAN PATH
 INPUT TYPE OF CHANNEL: 2

ENTER THE MULTIPATH SPREAD TIME (IN MICROSECONDS, F9 2):  0.500000E 03

ENTER THE DOPPLER SPREAD BANDWIDTH (IN HERTZ, F7.2):   1.00 HZ

ENTER THE SPECULAR-TO-MULTIPATH POWER RATIO (IN DB,  3):  20 DB
```

FIG. C.1.7

```
SET PARAMETERS FOR THE SECOND FADING CHANNEL:

TYPES OF FADING CHANNELS AVAILABLE:
 1: NO SPECULAR COMPONENT (RALEIGH FADING)
 2: SPECULAR COMPONENT, SHORTEST PATH
 3: SPECULAR COMPONENT, MEAN PATH
 INPUT TYPE OF CHANNEL: 2

ENTER THE MULTIPATH SPREAD TIME (IN MICROSECONDS, F9 2):  0.500000E 03

ENTER THE DOPPLER SPREAD BANDWIDTH (IN HERTZ, F7.2):   1.00 HZ

ENTER THE SPECULAR-TO-MULTIPATH POWER RATIO (IN DB,  3):  20 DB

ARE MULTIPLE SNR VALUES TO BE TESTED? Y-YES,N-NO: Y


INPUT THE RANGE OF SNR VALUES TO BE TESTED (IN DB):
INPUT THE INITIAL VALUE OF SNR (-99 -- +99, I3):-10  B
INPUT THE INCREMENT VALUE (01 -- 99, I2): 2 DB
INPUT THE ENDING VALUE OF SNR (-99 -- +99, I3):  40  B
IS SPACE DIVERSIFY RECEIVER REQUIRED? (Y/N)  Y
INPUT DURATION BETWEEN DECISION TIME FOR THE RECEIVE  (F7.5, IN SECOND)  0.00010


INPUT THE APPROXIMATE DURATION OF SIMULATION IN SECO DS (0.01 - 9.99, F4.2): 0.10 SEC.


TYPE OF PERFORMANCE MEASUREMENT AVAILABLE:
  1. COMPARE RECOVERED OUTPUT SIGNAL TO ORIGINAL INP T SIGNAL
  2. MEASURE OUTPUT SIGNAL TO NOISE RATIO

INPUT TYPE:2
```

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FIG. 6.1.7.

INPUT SNR (DB)	OUTPUT SNR (D)
-10.0000000	0.4142446
-8.0000000	0.3717036
-6.0000000	1.2635955
-4.0000000	1.8919516
-2.0000000	3.3436413
0.0	6.3902054
2.0000000	8.7668571
4.0000000	16.2037811
6.0000000	25.8732605
8.0000000	40.4527435
10.0000000	41.9621124
12.0000000	43.4101563
14.0000000	45.6305847
16.0000000	47.4060516
18.0000000	48.3931580
20.0000000	49.3137817
22.0000000	52.0993042
24.0000000	51.2066956
26.0000000	54.1014099
28.0000000	52.5127716
30.0000000	56.6009521
32.0000000	52.6348877
34.0000000	55.8170471
36.0000000	54.9163818
38.0000000	55.3856812
40.0000000	59.0605927

FIG. 6.1.8.

```
TYPE OF CALL AVAILABLE:
1: M1->M2, RURAL MOBILE TO RURAL MOBILE IN SAME UHF  EAM
2: M1->M3, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN SAME S-BAND
3: M1->M4, RURAL MOBILE TO RURAL MOBILE IN DIFFERENT UHF BEAM, IN DIFFERENT S-BAND
4: M1->FC1, RURAL MOBILE TO FIXED IN SAME S-BAND BEA
5: M1->FC2, RURAL MOBILE TO FIXED IN DIFFERENT S-BAN  BEAM
6: FC1->M1, FIXED TO RURAL MOBILE IN SAME S-BAND BEA
7: FC2->M1, FIXED TO RURAL MOBILE IN DIFFERENT S-BAN  BEAM

INPUT TYPE OF CALL TO BE SIMULATED: 1

MODE OF CALL AVAILABLE:
1: M1->M2, HARD WIRED TRANSPONDER
2: M1->M2, DIRECT SWITCHED TRANSPONDER
3: M1->M2, INDIRECT SWITCHED TRANSPONDER
4: M1->G1->M2, DOUBLE HOP SYSTEM

INPUT MODE OF CALL TO BE SIMULATED: 1
FREQUENCY OF THE BASEBAND SIGNAL ( LESS THAN 3000 HZ  ) (IN HERTZ, F7.2): 1000.00 HZ
NOTE : POWER OF THE BASEBAND SIGNAL HAS TO BE LESS O  EQUAL THAN .5 WATTS;
       SO THAT THE MAXIMUM INSTANTANEOUS FREQUENCY D VIATION IS
       LESS OR EQUAL THAN 12000 HZ
POWER OF THE BASEBAND SIGNAL(IN WATTS, F7.2):   0.50 WATTS
THE CARRIER POWER (IN WATTS, F7.2):   1.00 WATTS
THE FREQUENCY DEVIATION (IN HERTZ, F8.2): 12000 HZ.  (FIXED FOR NOW)

HOW MANY TIMES THE NYQUIST RATE DO YOU WANT THE SAMP ING FREQUENCY TO BE? (2 - 4, I1): 2
THE CARRIER FREQUENCY (IN HERTZ, F7.2):(NOT USED FOR NOW)

DO YOU WANT THE COMPRESSOR/EXPANDER IN?  Y-YES , N-N  : Y

DO YOU WANT THE PRE-EMPHASIS - DE-EMPHASIS FILTERS I  ?  Y-YES , N-NO : Y

DO YOU WANT A CO-CHANNEL INTERFERER IN THE UPLINK? Y YES,N-NO: N

IS FADING PRESENT IN THE UPLINK   CHANNEL? Y-YES,N-N : N

DO YOU WANT A CO-CHANNEL INTERFERER IN THE DOWNLINK? Y-YES,N-NO: N

IS FADING PRESENT IN THE DOWNLINK CHANNEL? Y-YES,N-N : Y

IS A SPACE DIVERSITY RECEIVER BEING USED? Y-YES,N-NO  N
THE DOWNLINK FADING CHANNEL IS PRESENT
```

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```
SET PARAMETERS FOR FADING CHANNEL.

TYPES OF FADING CHANNELS AVAILABLE:
 1: NO SPECULAR COMPONENT (RALEIGH FADING)
 2: SPECULAR COMPONENT, SHORTEST PATH
 3: SPECULAR COMPONENT, MEAN PATH
 INPUT TYPE OF CHANNEL: 2

ENTER THE MULTIPATH SPREAD TIME (IN MICROSECONDS, F9 2):  0.500000E 03

ENTER THE DOPPLER SPREAD BANDWIDTH (IN HERTZ, F(.2):   1.00 HZ

ENTER THE SPECULAR-TO-MULTIPATH POWER RATIO (IN DB,  3):  10 DB

ARE MULTIPLE SNR VALUES TO BE TESTED? Y-YES,N-NO: Y
```

FIG. 6.1.8.

```
 INPUT THE RANGE OF SNR VALUES TO BE TESTED (IN DB):
 INPUT THE INITIAL VALUE OF SNR (-99 -- +99, I3):-10  B
 INPUT THE INCREMENT VALUE (01 -- 99, I2): 2 DB
 INPUT THE ENDING VALUE OF SNR (-99 -- +99, I3):  40  B
 IS SPACE DIVERSIFY RECEIVER REQUIRED? (Y/N)  N


 INPUT THE APPROXIMATE DURATION OF SIMULATION IN SECO DS (0.01 - 9.99, F4.2): 0.10 SEC.


 TYPE OF PERFORMANCE MEASUREMENT AVAILABLE:
   1. COMPARE RECOVERED OUTPUT SIGNAL TO ORIGINAL INP T SIGNAL
   2. MEASURE OUTPUT SIGNAL TO NOISE RATIO

 INPUT TYPE:2
```

FIG. 6.1.8.

INPUT SNR (DB)	OUTPUT SNR (D)
-10.0000000	0.3327452
-8.0000000	0.4601035
-6.0000000	1.3240013
-4.0000000	3.0389242
-2.0000000	3.9141635
0.0	5.3271914
2.0000000	8.9695168
4.0000000	12.2795773
6.0000000	16.6423833
8.0000000	24.9819031
10.0000000	41.6535492
12.0000000	41.2625885
14.0000000	37.7913971
16.0000000	45.8797760
18.0000000	46.1225044
20.0000000	47.6090393
22.0000000	42.5114441
24.0000000	47.1889191
26.0000000	47.5585327
28.0000000	46.8480682
30.0000000	46.5578918
32.0000000	49.9220734
34.0000000	49.4653778
36.0000000	44.8569489
38.0000000	49.0057526
40.0000000	50.4644012